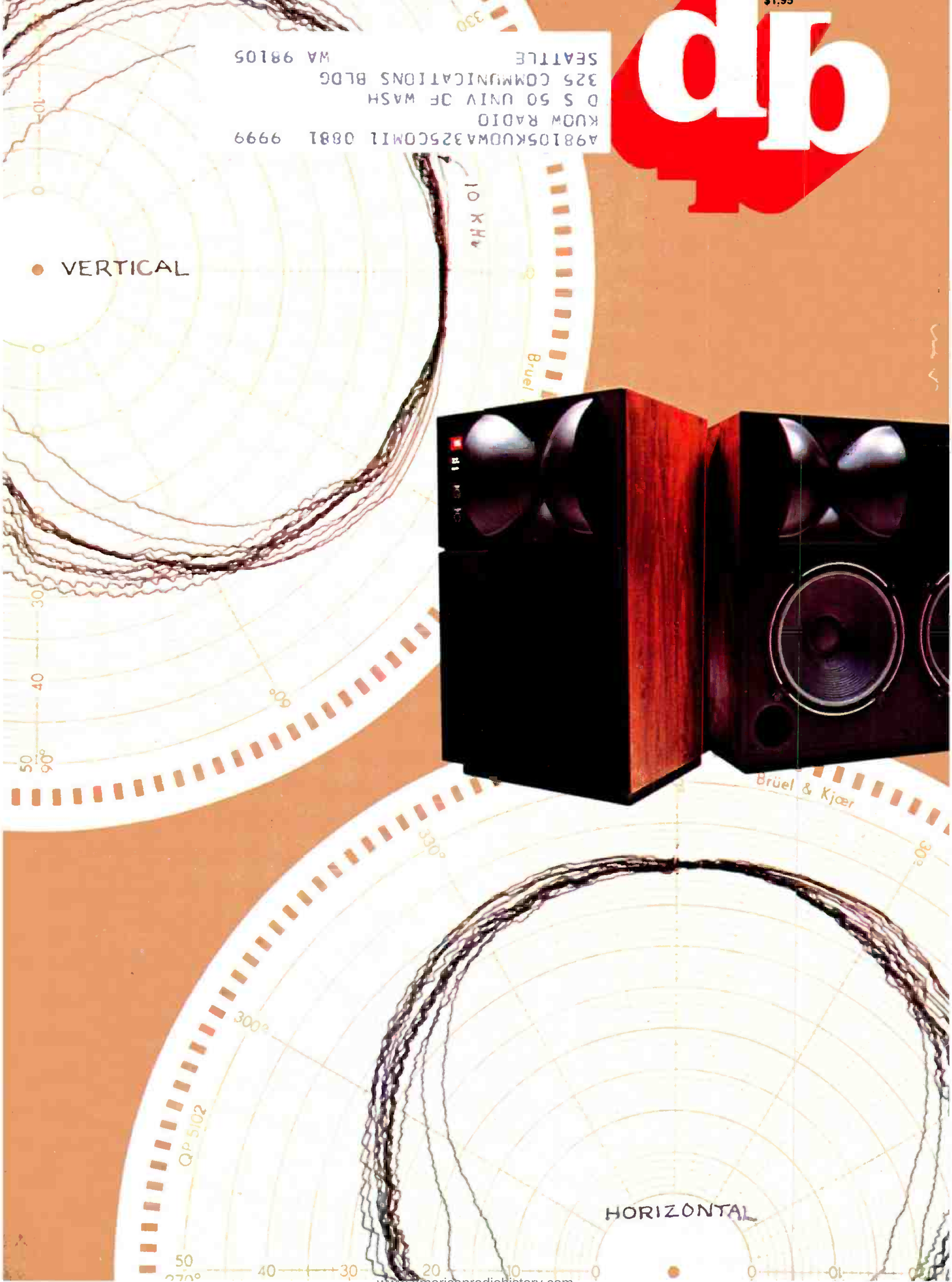
THE SOUND ENGINEERING MAGAZINE

AUGUST 1981
$1.95

db




SHURE
SM 85

Publisher
Larry Zide

Editor
John M. Woram

Associate Editor
Mark B. Waldstein

Advertising Production & Layout
Kathy Lee

Classified Advertising
Carol Vitelli

Book Sales
Lydia Calogrides

Circulation Manager
Eloise Beach

Art Director
Bob Laurie

Graphics
K&S Graphics

Typography
Spartan Phototype Co.

sales offices

Roy McDonald Associates, Inc.
Dallas, Texas 75237
First Continental Bank Bldg.
5801 Marvin D. Love Freeway
Suite 303
(214) 941-4461

Denver, Colorado Area
Englewood, Colorado 80112
14 Inverness Dr. East
Bldg. 1—Penthouse
(303) 771-8181

Houston, Texas 77036
6901 Corporate Drive
Suite 210
(713) 988-5005

Los Angeles Area
Glendale, California 91204
424 West Colorado Street
Suite 201
(213) 244-8176

Portland Area
Hillsboro, Oregon 97123
510 South First
P.O. Box 696
(503) 640-2011

San Francisco Area
Emeryville, California 94608
265 Baybridge Office Plaza
5801 Christie Avenue
(415) 653-2122

Sagamore Publishing Co., Inc.
New York
Plainview, NY 11803
1120 Old Country Rd.
(516) 433-6530

db

THE SOUND ENGINEERING MAGAZINE
AUGUST 1981 VOLUME 15, NO. 8

DEPARTMENTS

FEATURES

ABOUT THE COVER

• Unveiled at last May's AES Convention, The JBL 4430 and 4435 Bi-Radial Studio Monitor bring constant coverage technology to studio monitor design. As shown by the compiled polar response curves, the monitors provide flat response over a wide range of horizontal and vertical angles.

db, the Sound Engineering Magazine (ISSN 0011-7145) is published monthly by Sagamore Publishing Company, Inc. Entire contents copyright © 1981 by Sagamore Publishing Co., 1120 Old Country Road, Plainview, L.I., N.Y. 11803. Telephone (516) 433-6530. db is published for those individuals and firms in professional audio-recording, broadcast, audio-visual, sound reinforcement, consultants, video recording, film sound, etc. Application should be made on the subscription form in the rear of each issue. Subscriptions are $15.00 per year ($28.00 per year outside U.S. Possessions, $16.00 per year Canada) in U.S. funds. Single copies are $1.95 each. Editorial, Publishing and Sales Offices: 1120 Old Country Road, Plainview, New York 11803. Controlled circulation postage paid at Plainview, NY 11803 and an additional mailing office.

db Letters

UTILIZING DIGITAL AUDIO IN COMMERCIALS

To the Editor:

I was quite pleased to read about what is "believed to be the first network spot to utilize digital audio" (db. May 1981—"People/Places/Happenings").

It reminded me of my first digitally recorded commercial for 3M Corporate on September 12, 1979, at The Record Plant in L.A. The commercial entitled "The First Time" spotlighted singer Laura Brannigan during a recording session utilizing 3M equipment, and further points out the constant innovations pioneered by 3M.

After appearing on network TV during the Kissinger specials, The NBC Evening News and several other shows, the commercials were contenders for the "Clio" awards.

It is certainly anticipated that if digital is to be utilized in commercials, manufacturers of TV sets join the ranks of those of us in the audio field who strive for better, cleaner and more positive marriage of this audio-visual system. We are now seeing this progress manifest itself somewhat in the motion picture field.

Fred Weinberg
President
Fred Weinberg Productions, Inc.

REVIEWING THIRD ALTERNATIVES

To The Editor:

Crowhurst's discussion (June 1981) of possible "third alternatives" to some of the issues in the field of hearing are most appropriate. Many arguments do tend to become polarized to two opposing dogmas because of the attitude of some alleged scientists to the effect that, "If I grant that your view is partly right, then I must have been partly wrong, and that cannot be." The sooner such childish behavior can be gotten rid of, the better it will be for all of us (but don't hold your breath).

I must point out, though, that neither of the "third alternatives" mentioned by Crowhurst is new. In regard to the question of whether perfect pitch is innate or acquired, Otto Abraham suggested 80 years ago that we may all be born with the ability to recognize specific frequencies, but that it is simply trained out of us (or at least it atrophies from disuse)

Index of Advertisers

Coming Next Month

- In September, as the NRBA holds its annual convention (see ad on page 63), we once again look in on broadcast audio. If you've always wondered how to mike a P.O.P. test, Mel Sprinkle's application note tells all. We'll also show two approaches to broadcast console designs as well as two approaches to signal processing. In one, WPLJ chief engineer Robert Deitsch describes an in-house design using off the shelf compression and crossovers. In the other, we present part two of Nigel Branwell's essay on dynamic processing.

because of the emphasis on *relative* pitch that characterizes musical performance. The weight of present evidence supports this alternative view more and more strongly.

The other "third alternative," leading to the hypothesis that loud *wanted* music produces less damage to the ear than loud *unwanted* music, is also not new, although perhaps not as ancient as Abraham's theory about perfect pitch. In 1968, Hormann had subjects perform a tracking task in which feedback was provided to the subject, not visually, but rather by means of a 95-phon noise. In half of the subjects, the noise indicated that they were on target, in the other half that they were off. The noise pattern was exactly the same for both groups, but there was no way that the subjects could discover this. The "bad noise" group (i.e., those for whom the noise indicated that they were off target) showed 18 dB of auditory fatigue, while the "good noise" group showed only 10 dB. So perhaps there is something to the notion that rock music is more hazardous to those who don't like it, even though most of us consider the idea somewhat preposterous. I strongly advise against uncritical extrapolation of this principle, however, until extensive corroboration of Hormann's results appears. That is, I hope that none of db's readers throw away their earplugs just because they *like* gunfire, chain saws, or power mowers—or even 115-dBA *classical* music.

W. Dixon Ward
Professor
University of Minnesota

REFERENCES

Abraham, O. Das absolute Tonbewusstsein. Sammelbde, int. Musikges. 3, 1-86 (1901).

Hormann, H. Larm—psychologisch betrachtet. Bild d. Wissenschaft 6, 785-794 (1968).

db Calendar

SEPTEMBER

25 **The Society of Broadcast Engineers 9th Annual Central New York Regional Convention.** Syracuse Hilton Inn. For more information contact: Convention Chairman Hugh Cleland, WCNY TV/FM, Liverpool, NY 13088. Tel: (315) 457-0040.

OCTOBER

7-9 **Natural Stereo Techniques for Recording Music Workshop.** University of Wisconsin at Eau Claire. For more information contact: Burton Spangler, Audio Coordinator, Media Development Center, UW—Eau Claire, WI 54701. Tel: (715) 836-2651.

13-15 **The 11th Conference of the Western Educational Society for Telecommunications.** Harrah's, Reno, Nevada. For more information contact: Dr. Donald Price, Media Production Services, California State University, Los Angeles, CA 90032. Tel: (213) 224-3396.

25-30 **The 123rd SMPTE Technical Conference Exhibit.** Century Plaza Hotel, Los Angeles. For more info contact: SMPTE, 862 Scarsdale Ave., Scarsdale, NY 10583. Tel: (914) 472-6606

NOVEMBER

12-15 **Billboard Magazine's 3rd Annual International Video Entertainment/Music Conference.** Beverly Hills Hilton, Los Angeles. For more information contact: Billboard Magazine's Conference Bureau, 9000 Sunset Blvd., Los Angeles, CA 90069. Tel: (213) 273-7040.

25-27 **Prosound '81 Professional Sound Equipment Exhibition.** West Centre Hotel, London. For more information contact: Batiste Exhibitions & Promotions, Pembroke House, Campsbourne Road, London N8. Tel: 01-340 3291.

BOSE

BARRY BLESSER

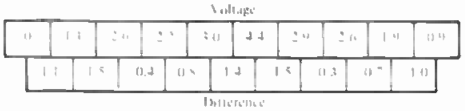
db Digital Audio

• We usually think of the digital representation of an audio signal as a numeric value, in digits, which represents an audio signal amplitude in volts. There is a one-to-one correspondence between these domains, which we might describe as being an *instantaneous, free form* transformation. This seems complex, but restated, it just means that each audio sample is independent and contains all of the information necessary to determine the digital word. The reciprocal is also true. The only restrictions on the transformation are a well-defined maximum for the digital peak, and a well-defined minimum corresponding to a single quantization level. The adjective "instantaneous" means that there is no memory in the system. Each sample does its own thing, without being influenced by any other sample.

We can, however, dream up other formats where one of these assumptions is not true. Consider a system in which the digital word specifies the difference between neighboring samples rather than the value of any single sample. With such a transformation, the audio voltages in FIGURE 1 would be represented by digital words corresponding to the difference values given in the figure.

If we transmit the differences instead of the actual values, the receiver can still determine the original audio values. For discussion purposes, assume that the transmitter knows that the first (reference) audio voltage is always 0.0. The first difference of 1.1 is transmitted and therefore the receiver knows that the first actual audio value is 1.1, since 0.0 (reference) plus 1.1 is just 1.1; the receiver determines the second audio value by taking the second difference (1.5) and adding it to the first derived value (of 1.1, to get 2.6); the third difference of −0.4 is added to the second derived value of 2.6 to get 2.2, and so on. Thus we see that the difference values are used to determine the actual values.

Figure 1. If the digital system specifies the difference between adjacent samples, it may be possible to transmit a digital signal with a smaller dynamic range.



However, notice that although the peak-to-peak audio range is 4.4, the peak-to-peak difference range is only 2.9 (1.4 maximum and −1.5 minimum). Note that the dynamic range of the difference is smaller than the actual signal's dynamic range. This suggests that the difference signal will require less bits to transmit without losing any information. This form of encoding is called *difference encoding*. If we connect an ordinary A/D converter to the difference signal, we create a technique called Differential Pulse Code Modulation (DPCM) instead of the simple Pulse Code Modulation (PCM).

PRE- AND DE-EMPHASIS

This should remind you of the familiar audio technique of pre- and de-emphasis as used in tape recorders and records. The encoded signal is a pre-filtered signal which is like the actual input signal except that the spectrum is changed. A differencing operation is like the pre-filter since the derivative operation is just a filter whose magnitude response is proportional to frequency. In fact, the differencing operation can be created by a simple differentiator made up of a capacitor and operational amplifier.

The analysis of such a system is very much like that of a tape recorder. If everything is designed properly, there is no effect on the signal since the pre- and de-emphasis cancel. Only the noise is affected since it is added after the pre-emphasis and is therefore only subject to de-emphasis. In digital audio, the noise is the quantization error. Such a system produces a white-noise spectrum which is filtered by an integrator. The integrator is the reverse of the differentiator. Clearly, there are many variations of a pre- and de-emphasis filter other than differentiating and integrating.

This kind of filter is especially interesting since it can be created digitally as well as by analog methods. In FIGURE 2, two versions of the differential PCM system are shown. The top version is the classical version with the A/D converter

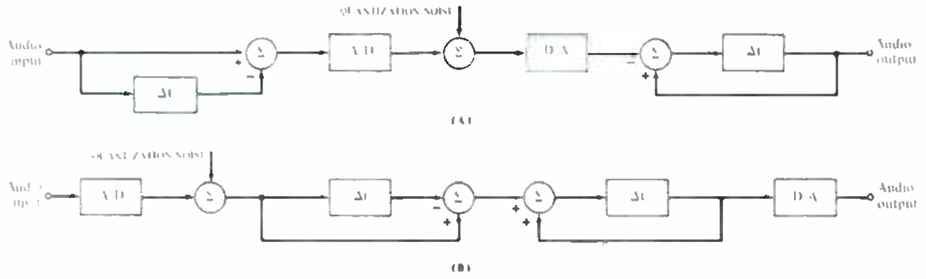


Figure 2. Two versions of a differential PCM system.

after the pre-emphasis differentiator. Actually, a difference filter is shown rather than a differentiator even though they are equivalent. At the subtractor, a delayed signal is subtracted from the main signal. This means that the result is the difference between the present value and the delayed value. When the delay is equal to the sampling period, the signal at the A/D is equivalent to that which would be obtained by subtracting the difference of two samples. At the receiver side, a summing unit is used to create the restored signal. Assume that its output is the present value of the audio signal. If the input is the difference, then the sum of the difference and the present value gives the next value. This process repeats. You can use the table of numbers as an example to demonstrate that the reconstruction really works.

From an audio point of view, this form of pre- and de-emphasis is not particularly interesting; however, FIGURE 2B shows a more interesting implementation. In this example, the differencing operation is performed *after* the A/D converter. The subtraction is also performed digitally. Thus, the pre-emphasis and de-emphasis are both performed digitally, with no source of noise between them. There is complete cancellation. Therefore, the quantization noise now passes through the system unchanged since it passes through both filters. Previously the noise was introduced after the pre-emphasis. This is called a lossless transform, since the digital pre- and de-emphasis transmits the difference signal without any loss of information. The interesting property of difference encoding is that peaks signal can become very small. Consider a 50 kHz sampling frequency having a 20 μsec. interval between samples. With a 100 Hz signal, the dynamic range of the difference signal is more than 40 dB below the actual signal. A 100 Hz sine wave does not change value very quickly. If the transmission word has 10 bits, then the audio signal can correspond to 20 bits. In other words, fewer bits are required to transmit a large signal. One can even get more dynamic range from a 20 Hz signal, since the derivatives are 5 times smaller. Higher frequencies do not do so well; and very high frequencies do much worse than simple PCM.

FIGURE 3 shows some of the properties of these systems. With analog pre-emphasis, the noise spectrum is shaped like a 1/f curve labelled *A* in the figure. This increases the noise at low frequencies (where the gain becomes very large) but decreases the noise at high frequencies. The digital difference encoding (B) does not change the noise with frequency. If one were to compute the noise power under the two curves, one would find that they are the same. We have merely traded one form of noise for another.

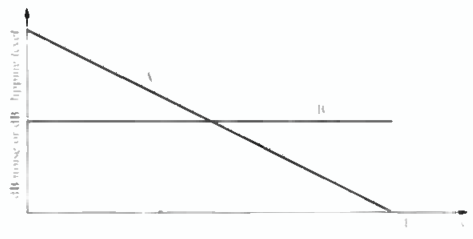
The same curves may be used to illustrate the clipping level for ordinary PCM and our DPCM. At low frequencies, curve A shows that the DPCM is much better than the PCM (curve B), but at high frequencies it is worse. Again, the area under the two curves is the same when viewed on a linear frequency scale.

These are only a few examples of how the information in a digital audio system can be transformed to allow the capacity of the system to be changed in one area by giving up performance in another.

OVERSAMPLING

Another variation of these systems can be created by increasing the sampling frequency to a much higher value, e.g. 500 kHz instead of 50 kHz. In this case, all audio signals may be considered as relatively low frequencies. The higher sampling rate means that the audio signal can not change value very much in between samples. At 500 kHz, the time between samples is only 2 μsec. In this short interval, all differences are extremely small. This decreases the transmitted dynamic range data to a very small value. Hence, only a few bits are required to represent the signal. This allows us to trade off the number of words per second (increased)

Figure 3. The same curves may be used to compare analog (A) to digital (B) noise, or, DPCM (A) to PCM (B) clipping level.



Circle 20 on Reader Service Card

Take a close look at what Tape Transports are going to be like in the '80s

The all new Telex 3000 is here NOW!

No industry has witnessed more technological improvements over the past few decades than our own. So, if you're looking for a tape transport for broadcast, studio or industry, it's important to choose a product with all the lastest industry innovations in one unit, the NEW Telex 3000. Write for exciting details about these features.

- Interchangeable head blocks
- A.Q.R. (Auto Cue Release)
- Differential electro-mechanical braking
- Spill-proof logic
- Motion sensing
- Tape counter
- 4 head capability
- Hyperbolic tape path
- Remote control capability
- Play only or record/play combinations
- Auto cue/rewind/cycle
- Dual speed
- Hysteresis drive motor
- Two torque spooling motors
- Quiet operation
- 120/240 volt operation
- Urethane pressure roller

Quality products for the audio professional

TELEX®

TELEX COMMUNICATIONS, INC.

9600 Aldrich Ave. So., Minneapolis, MN 55420 U.S.A.
Europe: 22, rue de la Légion-d'Honneur, 93200 St. Denis, France

Circle 23 on Reader Service Card

for the number of bits per word (decreased). Of course, we require converters to work at a much higher rate, but the problem of anti-aliasing filters is made trivial since the sampling frequency is so high.

The limiting case of such a system happens when the differences become so small that the digital word reduces to one bit. This degenerate form of DPCM is called *Delta Modulation*. At each sample, the difference is merely represented as higher or lower. Consider that this one bit represents the information +LSB or −LSB. (We cannot represent zero change since this would require more than one bit. There is an inherent assymetry in the digital word since zero must use up either one of the positive values or one of the negative values. For ordinary PCM we do not notice this because there are so many levels but there is in fact one more negative value than positive value. Delta modulation gives up the zero value since we need both a positive and a negative value.)

FIGURE 4 shows how one would build a digital delta modulation system. The audio signal is used to create the one-bit word represented as x in the figure. This value is either +1 or −1. The delay and summing network is just the simple integrator which we discussed before. The last value of y is added to the present x to create the next value y. The y is compared to the audio input. If the audio is larger than y, the comparitor adds a +1 difference in the integrator; if the audio is smaller, it adds a −1. Since the integrator in the encoder is identical to the integrator in the decoder, the value of y' will be the same as the value of y.

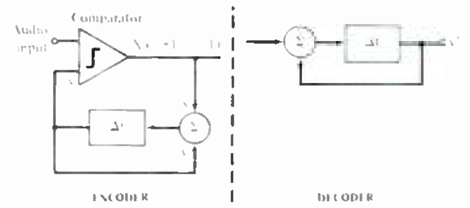


Figure 4. Block diagram for a delta modulation system.

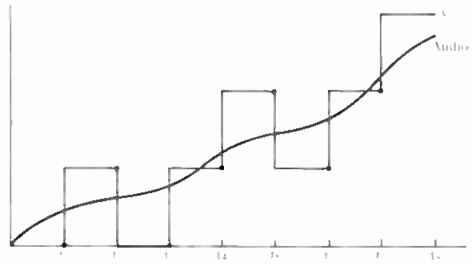


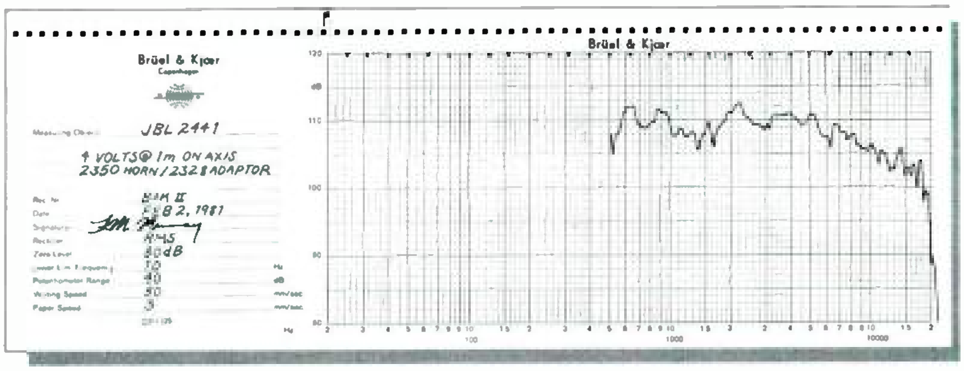
Figure 5. A simple low-pass filter will smooth out the delta modulation system's output (Y) to create a good representation of the audio input signal.

Since the value of y (or y') is being updated continuously, it will tend to track the audio signal. FIGURE 5 illustrates the process. At t_1, the value of y is less than the audio, so the comparitor adds 1. The resulting value of y at t_2 becomes larger than the audio, so the comparitor subtracts 1. At t_3, the comparitor adds 1. At t_4, y is again lower than the audio, so it adds 1 again. The curve y will follow the audio even though there are small steps. The rough quality can be ignored since a simple lowpass filter will smooth it out to create a good representation of the audio. Remember, a real PCM will also have steps in it when looked at carefully. You can think of each step as being an LSB.

This is perhaps the most simple A/D converter possible since the digitization can be realized by a simple one-bit comparitor costing 59 cents and there is essentially no anti-alias filtering required since the sampling frequency is extremely high. Alas, however, there is no free lunch. In the next article we will discuss all the ways in which you can get indigestion from using a delta modulation system. And as a finale, we will discuss various forms of digestive aids. This form of encoding justifies some considerations because of the extremely low cost. ■

(In the meantime, for more on delta modulation, see "Digitizing Audio with Delta Modulation" in the April, 1979 issue of db*—Ed.)*

This is not the only reason to buy JBL's new 2441 compression driver.



Unequalized frequency response of a typical 2441 on a JBL 2350 horn.

As you can see from this frequency response curve, the new JBL 2441 delivers impressive levels of performance. From extended bandwidth to high sensitivity and smooth, peak-free response.

But as important as these performance parameters are, they're only part of the story. Using the latest laser holography and computer analysis techniques, JBL engineers have developed a unique diaphragm design that allows the 2441 to match its outstanding response with unprecedented reliability and power capacity. That means you get exceptionally high performance without the trade-offs found in previous driver designs.

The secret behind this increased performance lies in the diaphragm's three-dimensional, diamond-pattern surround.[1] As outlined in a paper published in the Journal of the Audio Engineering Society,[2] this surround is both stronger and more flexible than conventional designs. This permits the diaphragm to combine all the traditional reliability and power capacity benefits of its aluminum construction with the extended frequency response of more exotic metals. It also maintains consistent diaphragm control throughout the driver's usable frequency range to eliminate uncontrolled response peaks.

Additionally, each 2441 is built to JBL's exacting standards. The magnetic assembly is machined from rugged cast iron and steel. Extremely tight machining tolerances and hand tolerance matching maintain unit to unit consistency. And finally, each 2441 is individually tested to ensure that it meets published specifications.

So before you buy any compression driver, ask your JBL professional products dealer about the 2441. It'll deliver a lot more than just an impressive frequency response.

Specifications		
Horn Throat Diameter	50 mm	2 in
Nominal Impedance	16 Ω	
Power Capacity	70 W continuous program	
Sensitivity (1 watt, 1 meter)	111 dB SPL (on axis of a JBL 2350 90° radial horn)	
Frequency Range	500 Hz to 18 kHz	
Voice Coil Diameter	100 mm	4 in
Voice Coil Material	Edgewound aluminum ribbon	
Flux Density	1.8 T (18,000 gauss)	

1. Patent Applied For

2. Journal of the Audio Engineering Society, 1980 October, Volume 28 Number 10. Reprints available upon request.

James B. Lansing Sound, Inc.
8500 Balboa Boulevard,
Northridge, California 91329 U.S.A.

JBL's diamond suspension diaphragm combines performance with reliability.

JBL Professional Products Division

Available in Canada through Gould Marketing, Montreal, Quebec

Circle 18 on Reader Service Card

Sights & Sounds

• Another Summer CES has come and gone and, as usual, everyone I come in contact with after this semi-annual event poses the same question: "What did you see hear that was new and different?" In terms of *product*, very little! In terms of *trends*, a great deal. Let's take a look at the most important product displayed at CES and see how it is likely to affect both audio and video in the years ahead. The product is 12 centimeters in diameter and has been dubbed the C-DAD, or Compact Digital Audio Disc.

The small disc is a dedicated audio disc, designed strictly for audio information storage, and not as an offshoot of a video disc system. Therein lies the controversy between two growing camps. There are those who maintain that the consumer of the mid-eighties (that's when digital audio is likely to take off—regardless of which format ultimately dominates the scene) is not about to buy yet *another* turntable for video discs and a separate player for digital audio. Those who subscribe to this argument insist that the same player should be able to handle both video discs and digital audio discs.

Countering these arguments are the co-inventors and co-developers of the C-DAD, Philips of Holland and Sony Corporation of Japan who, a couple of days before SCES, held a press conference in New York in which they revealed details concerning the small digital audio disc and their marketing plans for it. Their presentations also highlighted several good reasons for favoring a dedicated audio disc. Not the least important of these was a reminder that the world employs *three* popular TV broadcast systems (NTSC, PAL and SECAM) and that if an audio digital disc were to be tied to a video disc system, there would then have to be *three* versions of each digital audio disc—an inventory situation somewhat reminiscent of the short-lived days of quadraphonic sound whose demise is often attributed to that very type of non-standardization.

A second argument in favor of the C-DAD has to do with its format and its physical configuration. To begin with, the Philips-Sony C-DAD is an optical-laser tracking system. Thus, there is no contact between the "pick-up" (in this case, a semi-conductor laser) and the surface of the disc. That fact, plus the small size of the disc, suggests the possibility of a new program source for car stereo systems which would not be practical with larger discs or with discs that require physical contact between pickup and disc surface, such as the small grooved digital audio disc developed by Telefunken and now considered to be out of the running.

It is interesting to note that the three digital disc systems which have thus far been proposed for audio applications parallel the three types of video disc systems which are either being offered for sale now or will be offered in the very near future. The small, optical-laser Sony-Philips system, of course, uses the same pickup principle as the Magnavox/MCA/Pioneer "Laservision" disc which has been available to consumers for some time. The groove-type disc developed by Telefunken of West Germany, while not identical in concept to the RCA Selectavision video disc, does require a stylus riding in a spiral groove. And, finally, the grooveless capacitance-pickup disc proposed by JVC is completely analogous to that company's video disc proposal which is known as the VHD system (the audio version is called AHD, for Audio High Density).

CHARACTERISTICS OF THE COMPACT DIGITAL DISC

Digital quantization format for the C-DAD is 16-bit linear, per channel and a sampling rate of 44.1 kHz is used—high enough to insure ruler-flat response out to 20 kHz. Its error correction scheme is capable of correcting drop-outs of as long as 3,548 bits, which corresponds to about 2.4 millimeters of linear travel of the optical pickup relative to the spinning disc. Incidentally, C-DAD revolves at a variable rate—approximately 200 rpm near the outer circumference and about 500 rpm at the innermost point (50 millimeters from the center of the disc). Play is from inner circumference to outside and rotation is *counter*clockwise.* Playing time of the disc is 60 minutes per side, in stereo, and it would be possible to record four

Prototype digital disc players by Sony (left) and Philips, exhibited at recently held Summer CES in Chicago. Disc itself has about 1/6th the area of conventional LP disc shown behind it, but plays for one full hour per side.

channels of information on a side with playing time cut in half. It is also possible to "sandwich" two active sides together to form a double-sided disc, but present indications are that the software companies who have already opted for the system will probably offer one-sided discs, initially at least.

An hour-long C-DAD recording would contain up to six *billion* bits of information in the form of tiny "pits" along a helical track. Each pit is about 0.6 microns in width and about 0.2 microns deep. These pits, and the flat areas between them, represent the "1s" and "0s" of the required binary language encoding which in turn represent the instantaneous amplitudes of the waveform that has been recorded. A semiconductor laser beam picks up the pattern of pits and flats using a concentrated light beam which is many times thinner than a human hair. Scanning is at a rate of around 4.3 million bits per second. With no physical contact between the laser pickup and the disc, there is no record wear, even with repeated playings.

In addition to encoded audio information, it is possible to record a great deal of non-audio data onto the disc. For example, in demonstrations that I witnessed at SCES, the selection number, remaining playing time and the like were all displayed on a screen, as picked up and decoded from the demonstration discs themselves. Future discs might well contain song titles and much more, depending upon how sophisticated and how costly the disc players are made.

As for the audio performance of the disc itself, we have already noted that frequency response extends from 20 Hz to 20 kHz. Signal-to-noise and dynamic range are both better than 90 dB. Stereo separation is also in excess of 90 dB, while harmonic distortion, referred to peak signal level, is less than 0.05 percent. There is no measurable wow-and-flutter, since rotational speed is synchronized with a clock generator inside the player and is itself governed by information contained in the track of the disc.

INTRODUCTION AND TRANSITION

From what I was able to learn about marketing plans for the C-DAD, we in the United States are not likely to see players until early 1983. Both Sony and Philips plan to introduce players in their own domestic markets (Philips in Europe, Sony in Japan) in the fall of 1982, and these introductions will be accompanied by the introduction of a wide variety of software (by Polygram in Europe, CBS/Sony in Japan, and possibly other record companies who will climb aboard the C-DAD bandwagon in the interim). In view of the long delay before we see these new discs and players offered for sale, why did Sony and Philips choose this time to announce and demonstrate this remarkably superior audio technology? Most experts agree that the early announcement was designed to forestall any further debate about "which system" should become the world standard. At SCES, Marantz surprised visitors with their own prototype digital disc player. Additional companies have announced in favor of the Compact DAD, including Matsushita Electric (whose familiar brand names include Panasonic, Technics and Quasar), a surprising development when one recalls the corporate interrelationship between Matsushita and JVC, the inventors of the competing AHD/VHD Capacitance Pickup system. Still, there is always a certain amount of risk involved in announcing a radical new technological breakthrough well ahead of its commercial introduction. Some maintain that the recording industry, already suffering from a downturn, will be hurt further by the publicity garnered for the new digital discs, as uninformed consumers reduce purchases of conventional software in anticipation of the new discs and players.

Those recording studios that have amassed digital tape masters will probably find themselves in a fortunate position since the digital tape masters they are now using to produce "audiophile" type discs will also serve to create the true digital disc catalogs of the future. Those studios with vast libraries of analog master tapes may find little or no use for them when the world "goes digital" in a few years. But as for the fears about the obsolescence of the conventional vinyl disc, I think that such fears are over-exaggerated. The digital disc will make its way into homes slowly, I feel, and the analog LP discs which have been sold by the billions are not about to be discarded overnight, or even over a span of ten or more years. I'm not saying that they will *never* become collectors' items—only that it's a bit early to discard our collections of 12-inch LPs before we see which way the digital disc is really going to go—and how soon it's going to get there. ■

EVM™ by Electro-Voice
The low-frequency loudspeaker

Electro-Voice EVM™ Series II loudspeakers represent the ultimate in maximum efficiency, low-frequency speaker design. Years of experience, testing and refinement have resulted in a series of loudspeakers that are ideally suited for professional high-level, high-quality musical instrument and sound reinforcement systems.

Series II speakers incorporate many unique and innovative refinements that result in a loudspeaker that combines incredibly high power handling capability, efficiency and mechanical durability. All EVM's are conservatively rated at 200 continuous watts per EIA Standard RS-426A. This procedure is substantially more stringent than the more common continuous or "RMS" sine wave test, because it provides not only a 200-watt long-term stress (heat) but also duplicates mechanically demanding short duration program peaks of up to 800 watts which can destroy speaker cones and suspension parts.

EVM's are the ideal speaker for vented and horn-type enclosures. They are also featured in Electro-Voice's TL line of optimally-vented low-frequency systems. TL enclosure builder's plans are also available for custom construction, and each EVM data sheet contains the Thiele/Small parameters which allow you to predict the large and small signal performance in vented boxes.

For these and other reasons, not the least of which is an unmatched record of reliability, EVM's have been universally accepted by sound consultants, contractors and touring sound companies. When specifying a low-frequency loudspeaker, there really is no other choice. EVM – by Electro-Voice.

EV Electro-Voice
a gulton company

600 Cecil Street, Buchanan, Michigan 49107

In Canada
Electro-Voice, Div. of Gulton Industries (Canada) Ltd.,
345 Herbert St., Gananoque, Ontario K7G 2V1

Circle 34 on Reader Service Card

NORMAN H. CROWHURST

db Theory & Practice

Dimensions and Progress

When you're climbing a mountain, and don't have a map or a trail to follow, it helps to find a spot where you can see how you have gotten as far as you have, as a basis for determining how to proceed higher. So, pursuing the line we started in our previous column on the importance of basics, let us see how the whole business got started, from "sea level" so to speak.

For centuries, long before anyone thought about telephones, microphones, or even the telegraph, observers noted that electrical phenomena were of two major forms: electricity and magnetism. Much earlier, men like Sir Isaac Newton had quantized mechanical things like mass and force. But it was a long time before experimenters found a way to quantize their observations on electricity and magnetism.

There were forces which provided the first means of quantizing the phenomena. Bodies with electrical charges attracted or repelled one another: that was force. Bodies that were magnetized also attracted or repelled one another. And by this time, experimenters had found that electrical currents interacted in a way similar to the current in a magnetic field.

So we had what was called "static electricity," with properties somewhat similar, but apparently unrelated, to magnetism, and "current electricity," which was quite different from magnetism, but apparently related to it in some way. But, was static electricity related to current electricity? One would think so, but finding out in what way required some kind of breakthrough: progress in knowledge.

Measurements were made to find out how charges of static electricity behaved, how magnetism behaved, and how current electricity was related to magnetism. Both electrical charges and magnetized bodies exhibited forces, one toward another, dependent on the magnitude of both; doubling either one doubled the force. Without two charges, or two magnetized bodies, there was no force; one charge, or one magnetized body, produced no force.

Now, Newton had put dimensions on force. Force is measured as mass multiplied by the acceleration the force will

When he was only 16, Mick built a studio in his basement, which later became PCI Recording of Rochester, New York. In 1971, he met another Rochester resident named Chuck Mangione. The two have worked together ever since. Mick spent a year at Act One Studios in Buffalo, then returned to PCI, before his association with Mangione brought him to L.A. in 1975. Since then, he's been an independent engineer, working with people like Peter McIan, Cher and Lani Hall.

ON GEOGRAPHY

"The difference used to be that there were different players, different producers, different artists recording in different cities. The records out of New York were a little more hard-hitting, energetic kind of records. The records that came out of L.A. were a little smoother. Stylistically, there was a difference. Now, they're moving around, recording different parts of albums in different places. I can't tell anymore. Half the time, I see a lot of the people I know from rhythm sections in New York out here. And when I've been in New York, I've seen a lot of friends from out here there. So you can very easily be fooled into thinking something is done where it isn't."

ON STEREOTYPING

"The first year I was here, I had to work on a lot of demos for people for practically nothing to demonstrate that I could record something else besides what Chuck did, because that would give me a very limited amount of work. It took a little while to get out of that. So I had to give away a lot of time to prove it."

ON HONESTY

"I don't like second-guessing. I mean, 'Yes, well, will the public like this? Will they love this?' I can't tell. And I think that really few producers really can tell in advance if the public is going to love the record or not. I think the best thing you can do from everybody's standpoint—artist, producer, musicians—make an honest record that everybody involved with loves."

ON SPECIAL EFFECTS

"I really haven't heard anything new for quite a while. I think that was mostly all explored by George Martin and the Beatles in the '60s. Now, it's refinements on that, putting different things together, you know. I don't think that I have heard a new effect in years—a new specific sound."

ON TAPE

"3M's new formulations came out first, usually. As a matter of fact, I know that, because 206 came out before 406 did. 250 came out before 456 did. 3M's been a leader with new formulations on tape.... I think 3M is ahead. The difference in audio is minimal. The difference in durability is great. After several hundred passes through the machine, 250 still has more oxide left on it, which is a big advantage. I'm not plugging the tape to get the ad. I really am using it, finding it more rugged. That's my main point."

SCOTCH 250 RECORDING TAPE WHEN YOU LISTEN FOR A LIVING.

produce in that amount of mass. (Mass is a scientific word for weight, but different in that weight depends on gravity, whereas mass is a measure of the amount of matter upon which gravity acts, to manifest itself as weight.)

Thus, one pound of weight is a measure of force, as well as of weight, because it represents the force that would accelerate one pound, at the rate gravity would make it fall, if free to do so, which is an acceleration of approximately 32 feet-per-second-per-second. Acceleration is the *rate of change* of velocity. Velocity is measured as distance-per-unit time: feet-per-second, or miles-per-hour. Therefore, acceleration is the rate at which those feet-per-second, or miles-per-hour, are changing; per second, per minute, or in whatever unit of time you choose.

So, acceleration has the dimensions of distance divided by time squared. This may be written as L/T^2, or if you prefer, LT^{-2}. Now force is measured as mass (M), times the acceleration which that force will produce in that mass, or, force = MLT^{-2}

As we said, such force between charged or magnetized bodies is only present when both of them are charged or magnetized. And another observation shows that the force is inversely proportional to the square of the distance separating the charges, or the magnets. Taking this into account, given a known force, due to two charges or magnets acting on one another, the measure of the combined effect of the charges is $L^2 \times MLT^{-2}$, or ML^3T^{-2}.

Next, since that is the effect of two such charges or magnets, and is proportional to the product of them, we take the square root of that expression to arrive at what each charge, or each magnet, contributes to the overall force. That way, if there is only one charge, or only one magnet, there is no force, because it is something multiplied by nothing. So the dimension of a charge, or of a magnet's strength, is $M^{1/2}L^{3/2}T^{-1}$.

The early experimenters found out that current is a rate of change of charge, or a charge in motion, and that currents interacting produce forces like magnetism. This leads us into two definitions of current, according to which phenomenon we use to measure it.

In the first case (a rate of change of charge), we simply divide charge by time, making the dimension for current $M^{1/2}L^{3/2}T^{-2}$. In the case of magnetism, the force produced in a magnetic field is proportional to the length of current-carrying wire exposed to the field. Therefore, current has a reciprocal dimension of length, for the length of current-carrying wire in the stated field. So using this definition for current, its dimension is $M^{1/2}L^{1/2}T^{-1}$.

So we have two dimensional definitions for current. Now, if a wire carrying a current is moved in a magnetic field, a voltage is produced. This means that if we use the elctromagnetic definition for current, and multiply that by a velocity, $L\backslash T$, or LT^{-1}, we have the electromagnetic definition for voltage, which thus is $M^{1/2}L^{3/2}T^{-2}$.

Now, in terms of static electricity, when the distance between two charges is changed, the voltage and the force change, being proportional to distance. So the electrostatic definition of voltage is $M^{1/2}L^{3/2}T^{-1} \div L$, which reduces to $M^{1/2}L^{1/2}T^{-1}$.

Now we get to something more familiar to audio folks: resistance or impedance, the dimensions for which are voltage divided by current. Using the electromagnetic definitions for each, the dimensions for a resistance or impedance come down to L/T, or a velocity. And using the static definitions, the definition for resistance or impedance comes down to T/L, or the reciprocal of a velocity.

So far, we've talked about dimension—length, mass, time—without regard to units. The time used in experiments has always been the second. Length can be measured in various units, but the scientific unit is the meter. One of the difficulties in measurement was the fact that voltages in static electricity experiments could be millions of times as big as those in current electricity, while currents in the latter could be millions of times as big as those in static electricity. That was why it was so difficult

"Our Auditronics 720 combines recording studio quality with live broadcast flexibility,"

says Graham Simmons, Chief Engineer at Miami's WPBT — Channel 2. "Auditronics developed this 36 in — 16 out audio mixing console to give us all the EQ, reverb and signal processing we need for studio quality multi-track recording of our productions."

"When you're a national production facility like we are, you've got to have an audio signal path that's strictly state-of-the-art. For example, our Auditronics 720 preamp design is the latest generation and gives us the best signal-to-noise performance available. It allows us to do multiple generation dubbing and mixing without noise build up."

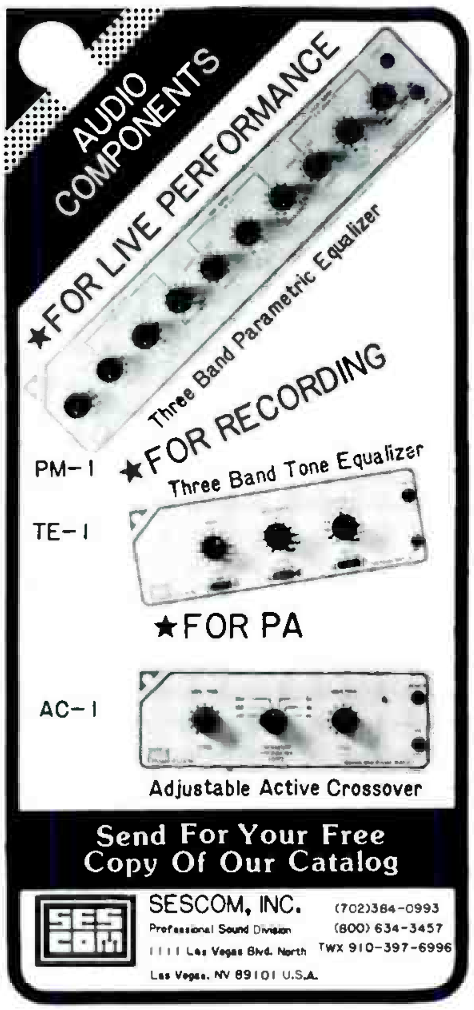
"In addition to its multi-track recording advantages, our 720 has the flexibility to do all the necessary mixing and signal processing in real time for a mono mix for TV, a stereo mix for FM simulcast, including network satelite feeds, as well as a scratch mix on videotape for later synchronization in post-production editing."

"Whether it's multi-track recording, live broadcast or post-production, our Auditronics 720 does everything we want it to do, and does it very nicely."

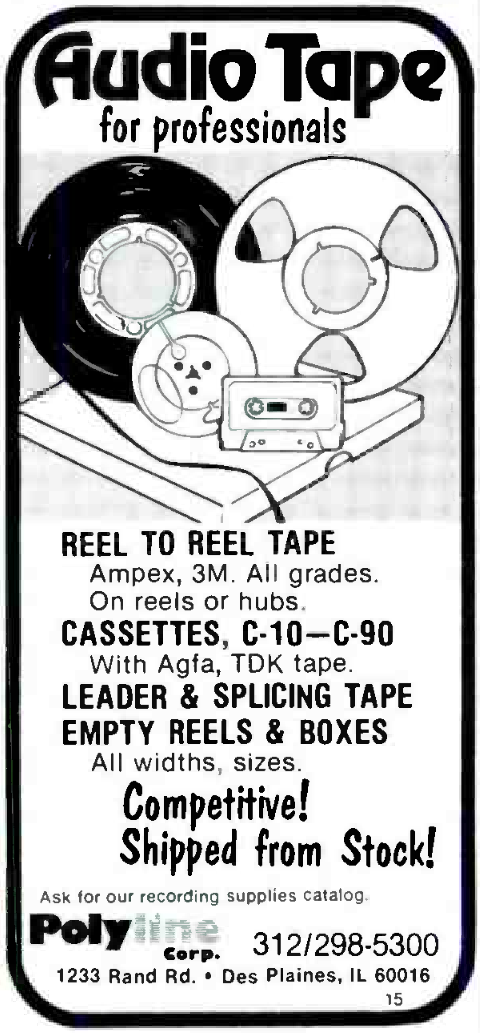
If you'd like to know what WPBT-Miami and over 300 other satisfied users know about Auditronics broadcast consoles, circle reader service number or contact:

auditronics, inc.
3750 Old Getwell Road
Memphis, Tennessee 38118
(901) 362-1350

Circle 19 on Reader Service Card



Circle 29 on Reader Service Card



Circle 25 on Reader Service Card

to interrelate the effects.

By the time very careful experimentation had enabled quantizing to be performed with both methods of measurement, it was found that the same resistance or impedance, measured in meter-gram-second units, had a value in static terms that had to be multiplied by 9 times 10^{16} to arrive at its value in electromagnetic terms. Now, since one has the dimensions of a velocity, and the other the dimensions of the reciprocal of a velocity, that ratio must be the square of a velocity, making the velocity 3 times 10^{8} meters per second.

Experiments in astronomy had already established that light travels at a little over 186,000 miles per second, which is very close to 300,000,000 meters per second. Could it be that light was a form of electromagnetic waves, and that other waves of the same form could be generated by electromagnetic means? That was what prompted experiments by Hertz, Marconi and others that eventually led to the invention of radio.

Further experiments, in the course of time, led to the development of polarized waves, synchronous reflectors using dipoles, waveguides of various types, and so forth: a whole new technology. The new technology is based on an understanding of electromagnetic waves in space, rather than on electricity and magnetism in conducting, insulating and magnetic substances.

Electromagnetic radiation is one form, consisting of two interacting elements of the wave, one electric, one magnetic, each in directions mutually at right angles to the direction in which the wave is propagating. These are called transverse waves. Acoustic waves, responsible for transmitting sound, are longitudinal. That is, the air particles vibrate back and forth along the same direction that the wave propagates in, rather than at right angles to it.

But both forms of radiation are waves, aren't they? They can transmit over distances that are relatively large (in terms of the dimensions used for each) because of mutually-sustaining properties that depend on their oscillatory nature: the fact that they have characteristic frequencies. They don't work in steady (dc) state, only when the quantities that vary, do so at some appropriate frequency. So, although electromagnetic waves are transverse, and acoustic waves are longitudinal, there should be some correspondences between them, shouldn't there?

In fact there are quite a few, if we will take the trouble to recognize them. For electromagnetic waves, an electric dipole serves as a reflector because it "shorts out" the electric component, doubling up the magnetic component. This means it re-radiates a wave in exactly opposite phase to the one causing those short-circuit currents. Behind the dipole (from the direction in which the wave arrives), the re-radiated wave cancels the original wave. But in front of it, the returning, re-radiated wave, will be in phase with the arriving wave, a quarter wavelength nearer the source so that a receiving dipole placed there will receive twice as much signal as the direct wave brings.

Synchronous reflectors occur in sound devices such as organ pipes. Sound travels along pipes and, before the invention of the telephone, this effect was used in the speaking tube. An organ pipe is a sort of speaking tube with either an open or closed end. When an arriving wave encounters an open end, the velocity of the air particular doubles and the sound pressure virtually disappears, resulting in a wave reflected back along the pipe.

When the arriving wave encounters a closed end, velocity is totally obstructed, so the pressure doubles up to compensate for the enforced zero velocity, and again a wave is reflected back along the pipe. With the open pipe end there is a pressure null at the open end, and at half-wave intervals, of whatever frequency builds up in the pipe, back along the pipe. With the closed end, there is a pressure null at a quarter-wave distance from the end, and then at half-wave intervals back from that.

Pipes will sustain a family of frequencies, having wavelengths that are, in the case of the open-ended pipe, a multiple of twice the length of the pipe. In the case of the closed pipe, the family of frequencies sustained will be such that the pipe is always an odd number of quarter-wavelengths long.

Notice somewhat of a similarity between electromagnetic waves and acoustic waves. Electromagnetic waves have two mutually self-sustaining elements: the electric and the magnetic. Acoustic waves similarly have pressure and particle velocity, or movement. In electromagnetic waves, the elements "move" mutually at right angles to the direction of propagation. In acoustic waves, the fluctuation occurs along the direction of propagation.

The electric and magnetic fields that make up those waves have a direction at right angles to the direction in which they are mutually propagated. On the other hand, the pressure in an acoustic wave can only fluctuate, at a given point: it has no direction, although the fluctuation can move in a direction. The particle velocity has a direction in which individual particles move, at a given instant and place, as well as a direction in which that component of the wave travels.

When the wave is truly longitudinal, which is the natural state for acoustic waves, the particle velocity is back and forth along the direction of propagation. But this is not always true, and the fact that acoustic waves are more complex than simply longitudinal, accounts for many of the effects used in acoustics that audio people need to be aware of. We will continue this next month. ■

1710A

00B 1701A

THREE GREAT WAYS TO CLEAN UP YOUR ACT.

The Sound Technology 1700 Series

Introducing three fast, accurate and easy-to-use Distortion Measurement Test Systems. All three deliver precise information for the audio professional. All three offer a multitude of features that make them indispensable tools for the engineer/technician.

Every system in the 1700 Series combines a flexible, ultra-low distortion sine-wave signal generator, a high resolution automatic-measuring THD analyzer, optional IMD analyzer and an accurate A.C. level meter in one instrument. Not only will a 1700 analyzer clean up your act, it'll clean up your bench too.

THE FEATURES

- Fully automatic nulling circuitry that measures distortion as low as .0009%. In less than five seconds.
- A simultaneously tuned, balanced and floating oscillator that's capable of 10Hz to 110kHz. With three digit pushbutton frequency selection that's both repeatable and fast. And, adjustable from −90 to +26dBm with three precision attenuators in 10dB, 1dB and .1dB steps (with ± 1dB vernier).
- You can measure signal-to-noise ratios with a 100dB dynamic range; floating sources from 30μV to 300 V; power in either dBm or watts; push a button and reject hum and high frequency noise with built-in 18dB/octave filters (30kHz, 80kHz and 400Hz).

WHICH ONE IS BEST FOR ME?

For most applications, the 1700B will do a big job for very few dollars. With the 1701A you get ultra-low residual distortion spec's, higher output level and integral, selectable meter detection circuits. Both models are great for R and D, engineering maintenance or Q.C. The 1710A was specifically designed for the broadcast and recording industry. It's the only distortion analyzer available with balanced and floating outputs and RFI shielding.

The 1700 Series Distortion Measurement Test Systems. Fully guaranteed for two years, parts and labor.

They're designed to be three great ways to clean up anyone's act.

Circle 35 on Reader Service Card

SOUND TECHNOLOGY
1400 Dell Avenue, Campbell, CA 95008
(408) 378-6540 Telex: 357445

Please send me more information on how the Sound Technology 1700 Series can help me clean up my act.

NAME ____________________
COMPANY ____________________
ADDRESS ____________________
CITY ____________ STATE ________
ZIP ____________ PHONE ____________

SOUND TECHNOLOGY Department 3008102

db Editorial

SPEAKING OF CONVENTIONS (as in last month's editiorial), it seemed for a time as though SPARS (the Society of Professional Audio Recording Studios) was ready to take up any possible slack in the convention circuit. For just as the AES announced its convention-cutback program, SPARS began publicizing its Conference '81, to be held in Nashville, Tennessee (about time *someone* woke up and discovered Music City, USA). A SPARS press release noted that "Exhibitors will have the opportunity to 'openly sell' ...to recording studio owners."

It occurred to some observers (or at least it did to us) that an aggressive SPARS "selling show" could really gain a foothold among manufacturers. Given the right conditions, a selling show geared specifically towards the industry's "big spenders" could prove almost irresistible to manufacturers.

But just as these words are coming off the typewriter, along comes word that SPARS has cancelled its 1981 convention, in response to "...growing industry-wide opinion that the industry is 'over-conventioned.'" Well, so much for our editorial on the timeliness of a selling show.

As an alternative, SPARS now plans a series of "road shows." At press time, details are still sketchy, but if our suspicions are correct, it could be that SPARS is having its own troubles trying to cope with the realities of the industry today.

It's easy enough to blame all of this on the sad economic state of the nation, the plight of the recording industry, and various other malevolent forces that cannot be traced back to our own doorsteps. But we also suspect that even in the healthiest of times, we do not need an endless sequence of conventions, punctuated by road shows and panel discussions in which blue-ribbon panels of experts sit around all night saying nothing in particular.

Maybe a tight economy will force all of us—the AES, SPARS, and even (gasp!) **db**—into a more-critical evaluation of the real needs of our industry.

We'd like to think that here at **db** we have all the answers, but as it turns out, we're having trouble enough trying to figure out what the questions are.

For example, on the subject of "road shows," just what does the industry really want? Or, should we ask, just what does the industry really *need*? (It might not be the same thing.)

The AES does an admirable job of presenting technical papers at its own road shows. We'd suspect the industry neither wants nor needs more of the same from others. (As an aside, the AES' convention cutback may even help raise the technical standards of the papers; with one less convention each year, session chairmen may be more selective in choosing papers.)

But the industry has other needs besides getting its quota of technical papers. There's a lot of high-technology out there that needs to be presented to the working studio engineer. It doesn't have to be spoon-fed, but neither can many of us digest the technical content found in some of the more-advanced work.

Here's where SPARS (and **db**!) can come in handy. Via "road shows" (and feature articles), the technology can be presented in a format that makes it accessible to the working audio pro'. It might not be as much fun as watching (or reading about) a recording studio "heavyweight" describing his life behind the big board, but, if you're trying to cope with an increasingly-complex industry, what would you rather have from us—amusement or education? Here at **db**, we're betting you'd choose the latter.

In this month's issue, John Hoge's feature article is a typical example of what we mean. There's probably more math here than many readers want to see (or so our typesetter assures us). Yet it's not of a kind that will present an impossible roadblock to any serious reader.

How many of you will lightly skim through the article and then rush out to design your own speaker system? Probably very few. But, how many will take the time to wade through the equations and come out with a better understanding of your own studio's system?

We hope that's where most of you are at today. And we'll even bet that many of you would go so far as to support a "road show" that brings today's technology a little closer to home. Although we'd like to think that **db** contains *everything* you really need to know (come to think of it, that's what the publisher tells our advertisers), in our more realistic moments we remember the value of attending all those conventions (and a few "road shows") over the past many years. We suspect that a strong "selling show" coupled with a sequence of intense workshops at the studio-operator level would meet with wide industry support. What do you think?

P.S. After a brief absence, our "Sound With Images" column is back this month, with noted columnist Len Feldman presiding. Len will keep us up-to-date on what's going on these days, as more and more image makers discover the creative potential of high-quality audio. However, we'll continue to remember that we're *not* a video magazine. In other words, Len's column will remain "SOUND With Images," not "*IMAGES!* (with sound)." ■

DEAN AUSTIN

Time-Aligned Loudspeakers Revisited

The ceramic magnet comes of age in professional monitor systems.

BACK IN THE "DARK AGES" (about four years ago), before we had come to accept double-digit inflation and the prime rate as a fact of daily life, there appeared a small dark cloud on the horizon, known in the loudspeaker world as a "scarcity of cobalt for making Alnico magnets." Without going into the political/economic reasons behind this scarcity, it actually did exist, and as a result the price of Alnico magnets was increasing at a rate that probably eclipsed even the price rise of crude oil. Of course, we had seen the production of consumer loudspeakers with ceramic magnets increasing steadily for about two decades, but in the field of professional studio monitors, the conversion and subsequent acceptance is much more recent.

A CERAMIC-MAGNET DUPLEX LOUDSPEAKER

At UREI, one of the earliest changes that occurred in the design of our 813 studio monitor system was made when the manufacturer of the auxiliary woofer (UREI 800W) sent us engineering prototypes of ceramic units for comparison with the Alnico magnet versions which were used for our initial production runs. After evaluating several samples of a ceramic-magnet replacement loudspeaker, and working towards improved sensitivity, a new ceramic-magnet 800W was procured that was completely interchangeable with the original version. But knowing that it would not be nearly so simple to change our Altec 604-8G duplex loudspeaker to ceramic magnets, we sincerely hoped that it would be manufactured forever.

Eventually, the cobalt situation became so tenuous that Altec informed us that a new ceramic-magnet duplex loudspeaker was indeed under development and a timetable should be established for converting our 800-family of monitor systems. There was at least one positive aspect to the use of an all-ceramic system. It is well known that Alnico magnets are subject to self-demagnetization when driven at very high levels of current through the voice coil (*what?... in a recording* studio?), whereas ceramic magnets are not so adversely affected by high drive levels.

As soon as engineering prototypes of the new custom-designed ceramic-magnet duplex loudspeaker were delivered, the fun began. The design objectives for our new model 813A were the same as those for the original 813 project:

1. flat frequency response (40 Hz to 16 kHz).
2. improved efficiency.
3. optimum damping.
4. low distortion.
5. time-alignment® of low- and high-frequency drivers.

In the final design of the 813A, it was again possible to meet all of these objectives. In fact, system efficiency is almost 1½ dB better in the piston band than the 813; another bonus provided by the ceramic magnets on the drivers.

As it turned out, most of the design effort was required to meet the first objective, with particular attention to that portion of the spectrum in the crossover region. (The development of the 813 enclosure, design considerations for the low and lower-mid frequency system response, and details of the time-alignment technique were covered in the author's "Time-Aligned Loudspeaker Systems" in the March, 1979 issue of **db**—Editor.) "Time-align" and its derivatives are registered trademarks of, and licensed by, E. M. Long Associates and the use of this terminology is licensed to UREI for use in connection with its products.

The tight bass sound has become one of the trademarks of the 800-series monitors. In the on-going 813A project, it was possible to effect some significant improvement in the damping at the low frequencies when measured by the tone-burst method. The correlation between tone-burst testing and subjective listening tests is most remarkable. The physical and mechanical changes required by the redesign, to accommodate the pancake-shaped ceramic magnet, produced some advantages through serendipity, but, conversely, created the need for other complementary modifictions in order to maintain smooth power response through the crossover region.

The power response of the ceramic low-frequency driver is indeed smoother in the 500 Hz to 1000 Hz region than the

Dean Austin is vice president of facilities & planning, United Recording Corp.

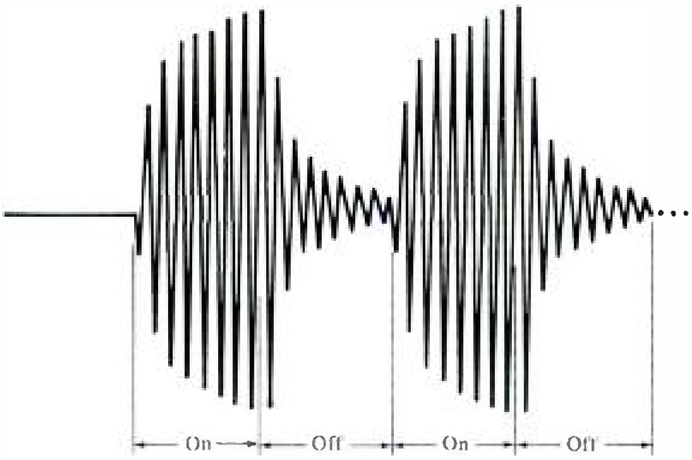


Figure 1. A tone burst at 2.3 kHz propagated through a horn with no buffer.

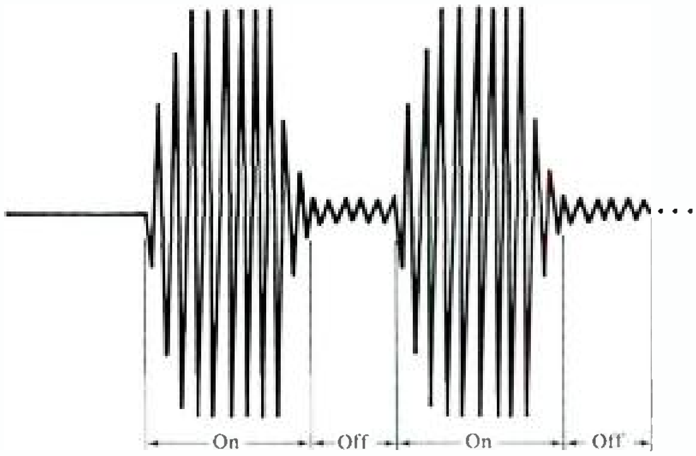


Figure 2. The same tone burst, after modifications were made to the horn.

original Alnico unit. However, the throat length and the diameter of the throat were considerably less than in the original 604-8G version. This required lengthening the horn and increasing the mouth area without producing a greater "shadow" effect, caused by the increased size of the high-frequency horn.

When it became apparent that most of the system engineering would be directed at the frequency spectrum above 1000 Hz, it was obvious that tests would be speeded up considerably by working in a small anechoic chamber instead of the "free-space" test tower used for the 813 project. Not having an anechoic chamber at hand, consideration was given to constructing one. At the May, 1980 Audio Engineering Society Convention in Los Angeles, UREI demonstrated the 800-series monitors in one of the hotel rooms. In an effort to make the room sound something like a typical control room, Alpha Audio made available to us some sheets of their new Sonex sound absorbing acoustic panels. The absorption of this material is very close to 100 percent in the audio frequency range above 1000 Hz. So, for a loudspeaker test facility, a small chamber was constructed by hanging 4-inch thick Sonex panels in a cubical framework and cutting a circular hole in one side to admit the test speaker, which was supported on a stand outside the "chamber." The test microphone was hung at a distance of one meter from the speaker and we were in business.

THERAPY FOR SOUND WAVES TRAUMA

It is an inescapable physical phenomenon that, with a short horn loading a high-frequency driver, such as is the case for all duplex loudspeakers with crossovers in the 1500-to-2000 Hz region, the transition which takes place at the end of the horn can be a traumatic experience for sound waves. The high-frequency audio waves travel through the throat of the driver after leaving the diaphragm, then propagate out through the horn, whose boundaries are almost 100 percent reflective. Then, they are suddenly dumped into the air. FIGURE 1 shows a tone burst at 2.3 kHz, propagated through a horn with no "buffer" at the transition boundary into the air. The burst is on for eight cycles and then off for eight, and the "ringing" which takes place when the burst is shut off is very evident. Also, there is a delay effect which causes the burst to build up gradually over the entire eight cycles, reaching a steady-state condition just as the burst is shut off again. As previously mentioned, work had been under way, before the change in duplex loudspeakers occurred, to devise some kind of a "diffraction buffer" which would reduce these transient distortions. This buffer would ideally be composed of a material with an optimized absorption coefficient, thus providing an "acoustic buffer" to match the reflective horn walls to the absorptive air. Measurement and experimentation with various materials led to the conclusion that a material with the proper absorption coefficients and optimum shape, would effect substantial improvement in performance. Since the horn length needed to be increased slightly, the diffraction buffer could accomplish both tasks with one piece of material. Considerable previous testing had established that a tear-drop shape worked well, so it was only necessary to determine the optimum dimension to properly load the high frequency driver down to the desired crossover frequency, and provide the desired buffering effect. Since the horn mouth is of two dissimilar materials, the increase in length required is not directly proportional to the crossover frequency change, but was determined by measurement.

As a result of testing up to this point, and from a study of available literature, it was believed that some transient distortion is introduced in a rectangular horn by multiple reflections along the junctions between its horizontal and vertical walls. One solution to this would be the use of a circular horn. But

Figure 3. Amplitude-versus-frequency response of the plastic horn.

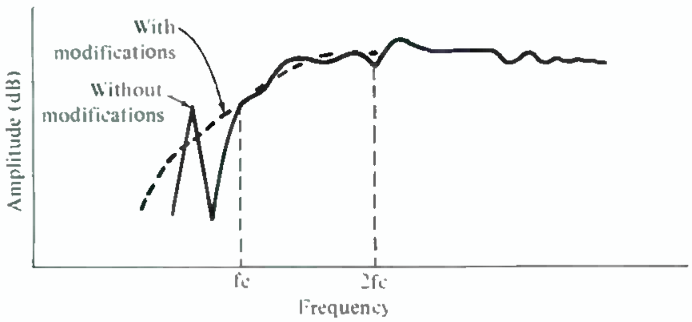


Figure 4. The modified horn, showing the diffraction buffer, absorption material and slot configuration.

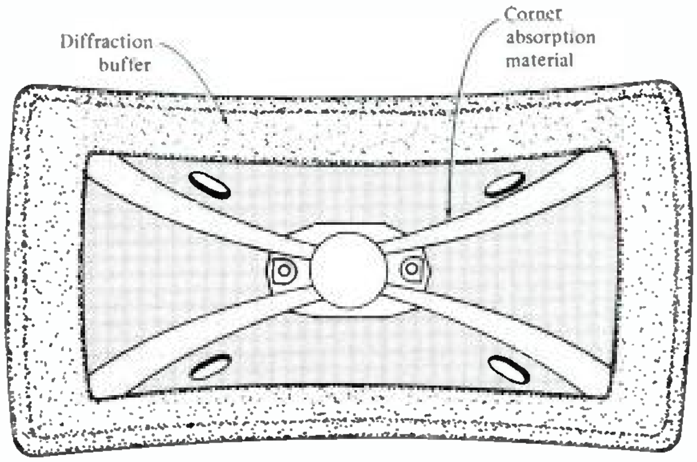


Figure 5. The completed UREI time-align 813A studio monitor system. The slots in the horn eliminate shadow effects caused by the horn's presence.

in a control room environment, a rectangular horn with an aspect ratio of about 2.5:1 gives much better concentration of the energy where it is needed.

SYNERGY

Here again came the benefit of the synergy that often occurs when an engineering project attacks several aspects of an overall design effort at once. In designing the high-frequency driver for their new duplex loudspeaker, Altec found it necessary to reduce the throat diameter to approximately seven-eighths of an inch. Although the flange supplied was still designed to mate with the custom UREI 800H horn, an impedance mismatch occurred in the throat which created phase shifts and consequently amplitude variations in the upper high-frequency response. The solution to these last two problems was found by forming a piece of foam material to fit inside the horn on its short vertical walls. Most of the surface of the material was covered with a highly reflective plastic membrane, the rigidity and mass of which is sufficient to provide a solid boundary at high frequencies. This did not reduce the efficiency of the system, but the edges of the material at the junctures of the walls in the horn absorb the corner reflections, eliminating that source of transient distortion, as verified by our measurements. FIGURE 2 shows the result of a tone burst applied to the high frequency section after all of the above described modifications were made to the horn. FIGURE 3 shows an amplitude response plot of the high frequency section with and without the modifications to the basic plastic horn.

THE SHADOW APPEARS

Although we had benefited greatly from all of the pieces of the puzzle falling into place up to this point, of course there had to be some compensating difficulty created somewhere along the line. Sure enough, it was found when the diffraction buffer was installed on the horn and a composite axial response of the duplex speaker was plotted. In any duplex-type loudspeaker there is some "shadowing" of the sound waves radiated near the center of the low frequency cone by the horn which is concentrically mounted in front of the woofer cone. When the horn is kept small enough, the effect is minimal, but with the new 800H buffer-diffracted horn, the shadow became apparent. Although the nondirectionality of low-frequency sounds makes it difficult to detect any blockage of the sound waves, phase differences occurring as the waves are diffracted around the horn do affect the frequency and power response.

The solution was to cut some holes in the horn to let the mid-frequency energy through. Considerable experimentation was required to optimize this slot configuration, but the mid-frequency response was restored to a flatness that defied detecting any "shadow effect" by cutting four slots in the horn as shown in FIGURE 4. The slots are approximately 1 cm-by-4 cm in size and are spaced well away from the vertical axis of the horn throat. By elongating the slots to a 4:1 aspect ratio, with the narrow side perpendicular to the direction of travel of the high frequency wave fronts, there was no measure degradation of the high-frequency response of the horn. Sufficient energy is radiated through the slots to eliminate the shadow created by the increased size of the high-frequency horn, which had resulted in the previously measured perturbations.

The development of the high-frequency horn (patent applied for) was the work of M. T. Putnam and the network for the 813A was the responsibility of Mr. Dennis Fink. The design of the 813A is an excellent example of producing a synergistic solution to a problem created by a change in design requirements, which in this case was precipitated by the necessity to change to a ceramic-magnet duplex loudspeaker. With the exception of the necessary network changes, most of the work was done with the high-frequency section of the system. Therefore, it was a relatively simple job to redesign the other members of the 800-monitor family, the 811A small system, and the 815A dual-auxiliary-woofer system. ■

SIDNEY L. SILVER

Digital Filters in Audio Signal Processing

A guide to the practical aspects of digital filtering.

AT THE PRESENT TIME, the full impact of digital filtering has not yet been felt by the audio industry. This has been due, mainly, to the high implementation cost and system complexity which, until recently, have severely limited the application of digital filter techniques in studio operations. However, recent advances in large-scale integration have made it possible to implement digital filter systems with just a few components, so that systems that once required numerous circuit boards can now be easily constructed with a small number of ICs. Also, power consumption has been drastically reduced in the new designs, and many of the elements of the filter can now be placed on a single LSI chip.

Digital filtering permits the efficient processing of analog data by transforming that data into a digital format before extracting meaningful information from the pulse stream. Although these filters are characterized differently from their analog counterparts, they are often required to perform identical operations. In either case, circuit components in the system function in such a way as to modify the frequency composition of an audio signal in a predetermined manner. However, for every problem encountered in the design of an analog filter, there is an equivalent one in the digital case. In analog, or continuous filtering for example, frequency stability depends upon the accuracy and matching of passive components. In digital processing, it depends mainly on the accuracy of the sampling frequency. Other error sources that must be dealt with in digital units are quantization noise arising from analog-to-digital conversion, and aliasing distortion caused by signal frequencies higher than half the sampling frequency.

One may therefore ask: What are the advantages of using digital filters in the audio field? And what properties do digital filters have that make them superior in performance to equivalent analog units? First of all, digital filters are insensitive to parameter drift; changes in temperature, voltage, and component values do not adversely affect digital elements the way they do analog types. Secondly, they can easily be designed with a frequency response closer to the ideal; that is, uniform passband response and infinite attenuation in the stopband. It is also possible to achieve extremely high Q values and steep cut-off rates (even at very low frequencies) that are difficult or even impossible to attain with conventional filters. With digital filters, there is virtually no limit to the signal-to-noise ratios that can be realized. All that is required is to increase the resolution of data bits at the filter input. There are also no impedance-matching problems and no insertion losses. Probably the most important feature is their great flexibility. For example, they can be easily modified by simple program changes, so that one filter unit provides many functions. Furthermore, using multiplexer techniques, digital filters can be time-shared with a substantial number of independent inputs, yielding an even more useful channel-bandwidth capability.

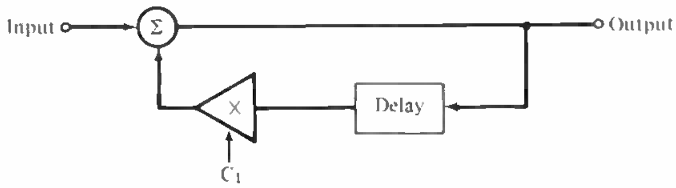
Unfortunately, most of the available literature on digital filters treats the subject primarily in mathematical terms, often based on idealized components. In this article, however, we shall restrict ourselves to the practical aspects of digital filtering, and discuss its use as a signal-processing element in audio work.

BASIC FILTER STRUCTURE

What is a digital filter? As yet, there is no universal agreement on a precise definition of the term. For our purposes, however, it can be defined as a process or device which operates on a coded sequence of binary digits produced by sampling a band-limited audio signal in time, and quantizing that signal in amplitude, so as to produce a new sequence with altered

Sidney L. Silver is on the supervisory staff of the Telecommunications Section of the United Nations, where he is in charge of sound and recording.

Figure 1. Basic circuit of a simple first-order digital filter.

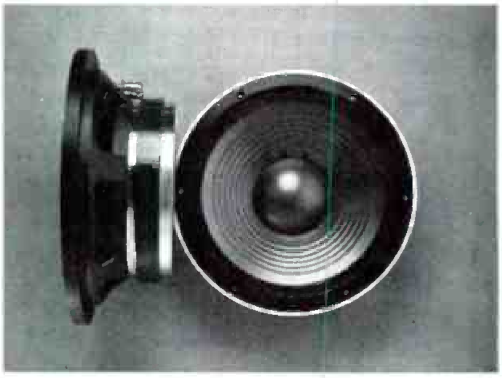


Experience previously unattainable performance in the mid-frequency range, an evolution of pure audio. Discover the best sounding 12" mid-frequency loudspeaker available. Listen... and experience the TM-1201 from TAD.

Assembled in the hands of veteran audio craftsmen, the TM-1201 is especially designed for mid-frequency applications, with a usable range of 200 to 3000 Hz. Ultra-low distortion, smooth extended frequency response and high power handling establish the TM-1201 as a hallmark in the evolution of sound.

Evolutionary in itself, with a lighter, more rigid material than currently used in the field, the polymer graphite cone has true pistonic motion throughout its specified range, with none of the peaks and valleys associated with cone break-up or doubling. In addition, a high power 4" voice coil with edgewound copper conductors better utilizes the flux within the magnetic gap to increase acoustic conversion efficiency. A corrugated cloth surround, for extreme linearity in high power applications, and a rugged 6-spoke die-cast aluminum frame assures both durable and reliable performance.

Conquer the peaks and valleys of sound with the unprecedented mid-frequency performance found in the TM-1201.



The evolution of sound begins with TAD.

TAD Technical Audio Devices

A division of U.S. Pioneer Electronics Corp.
142 Redneck Ave., Moonachie, N.J. 07074
(201) 440-8234 Telex: 133484 PIONERUSA MOON

Detailed specifications and performance data available upon request.

frequency content. After the pulse stream is processed, the filtered output may be left in its digital form for further processing by a digital computer, or converted back into its analog form, according to application.

To introduce the concept of digital filtering, let us consider the simple first-order filter (6 dB/octave-slope) shown in FIGURE 1. Here the basic building blocks are an adder, a delay line, and a multiplier. Note that there are none of the passive components (resistors, capacitors, or coils) associated with conventional analog filters. It can be seen that the filter output is fed to the delay line whose output, in turn, is applied to the multiplier. At the adder, the multiplier output is combined with the input. In this operation, the multiplier and adder comprise the arithmetic unit, or numerical calculator of the filter. The delay unit, usually in the form of a shift register or RAM (random-access memory), compares the data sampled at a particular instant with the data sampled previously.

The delay length is equivalent to one sampling interval, and we assume that the addition and multiplication operations are instantaneous. Since feedback is used in its implementation, the output of the filter is effectively derived from the present input values and from past values of the input. In this process, data is extracted sequentially from the delay element and passed to the multiplier which provides the scaling factor, or filter coefficient, that determines the frequency response. The scaling factor represents a digital number which, once selected, defines the fixed characteristics of the filter.

The output of a digital filter may be expressed by a difference equation which defines the output data value, y(nT), as a function of the present input data value, x(nT), and any prior input and output values. For a first-order filter, the general equation is:

$$y(NT) = x(nT) + Cy(nT-T)$$

where n is the number of sampling intervals at a given time, T is the period of sampling, and C is the scaling factor, or filter coefficient.

MAJOR FILTER TYPES

There are many ways of designating digital filters, but all of them fall into three main categories—recursive, transversal, and lattice. Recursive filters, which will be discussed first, use feedback in such a way that the filtered output is always an explicit function of past and present input values. These units are referred to in the literature as IIF (infinite impulse response filters).

FIGURE 2 shows a second-order recursive filter comprising two adders and four multipliers, with two shift registers serving as the delay elements. These components provide sufficient flexibility, so that by appropriate choice of scaling factors, the filtering action can be made equivalent to virtually any corresponding analog type. Here the set of filter coefficients, C_1 through C_4, are entered into a ROM (read-only memory), and fed sequentially to their respective multipliers. By merely

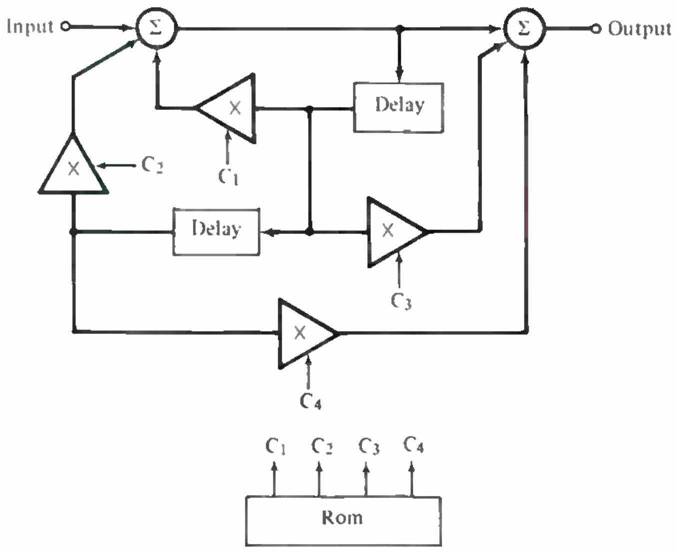


Figure 2. A second-order recursive filter with a 12 dB-per-octave slope. C_1-C_4 represent a set of filter coefficients.

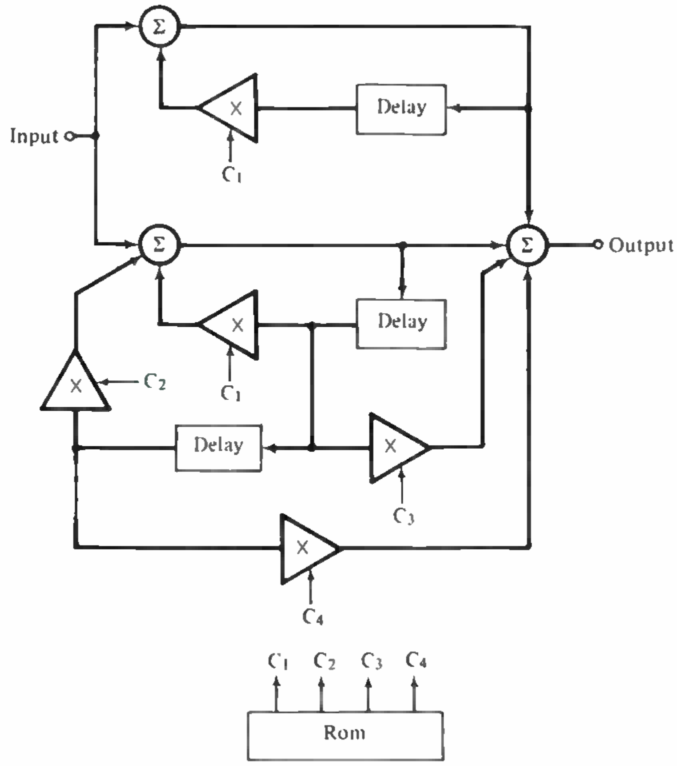
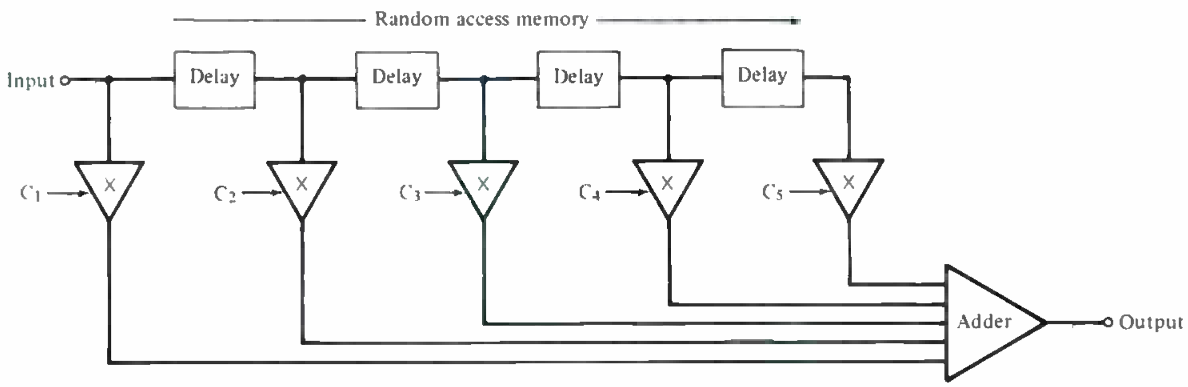


Figure 3. A third-order recursive filter is derived by placing a first-order and a second-order section in parallel.

Figure 4. Implementation of a fourth-order transversal filter.




AGFA
sound-sational!
AGFA has been answering the high level requirements of a demanding professional U.S. and European market for many years. And surpassing it everytime – with quality, performance, reliability and availability.
Our Studio Mastering Tape has been accepted on the highest level as "the tape" for top quality original recording. It's available to the discerning in a low-noise, high-output, low print tape in ¼", ½", 1" and 2".
When it comes to sound – come to AGFA for Studio Mastering Tape. It's sound-sational!
Studio Mastering Tape
AGFA-GEVAERT, INC.
275 North Street, Teterboro, N.J. 07608
(201) 288-4100
Circle 26 on Reader Service Card

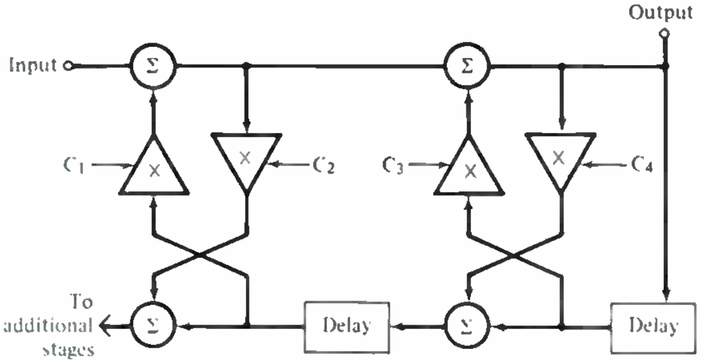


Figure 5. A second-order lattice type digital filter.

changing these scaling factors, any of the classic filter functions (Butterworth, Chebyshev, Bessell, elliptic, etc., in various modes of highpass, lowpass, bandpass, and bandstop) may be realized from the same filter unit. Also, by assembling any number of first-order and second-order filter sections in series, parallel, or a combination of both, it is possible to synthesize a wide variety of higher-order filter functions. As an example, FIGURE 3 shows how a third-order filter can be implemented by putting a first-order section and a second-order section in parallel.

In the transversal filter configuration, the output data values at any given instant depend upon the input values at that time and on the number of past inputs. Since there are no feedback elements employed, past inputs do not enter into the response. Transversal filters generate a finite impulse response and are therefore also termed FIR filters. FIGURE 4 illustrates a typical example of a fourth-order transversal filter requiring only a single adder and five multipliers, with a RAM (random-access memory) as the delay element. Clearly, the structure resembles that of a tapped delay line, each tap representing a fixed delay at uniform intervals related to the sampling time.

In operation, the filter produces a weighted running average of incoming data values. Each data value is successively multiplied by an appropriate filter coefficient stored in a ROM, and digitally summed in an adder at the output. This sequence continues for each sampling cycle as each data value advances in position along the delay line memory. The tapped delay weights are adjusted to give the sequential values in the impulse response that corresponds to the desired frequency response. An important advantage of transversal filters is their unconditional stability, which means they are virtually immune to environment changes. One limitation is that a large number of delay elements are needed to obtain steep cutoff rates.

The lattice type digital filter (FIGURE 5) is a relatively new form which offers a compromise between recursive and transversal units. It has greater stability than the recursive filter, but requires less hardware than the transversal type. However, it generally operates with larger coefficient sizes, making it more difficult to program than the other configurations. The full potential of lattice filters in audio applications has yet to be explored.

TIME-VARYING FILTERS

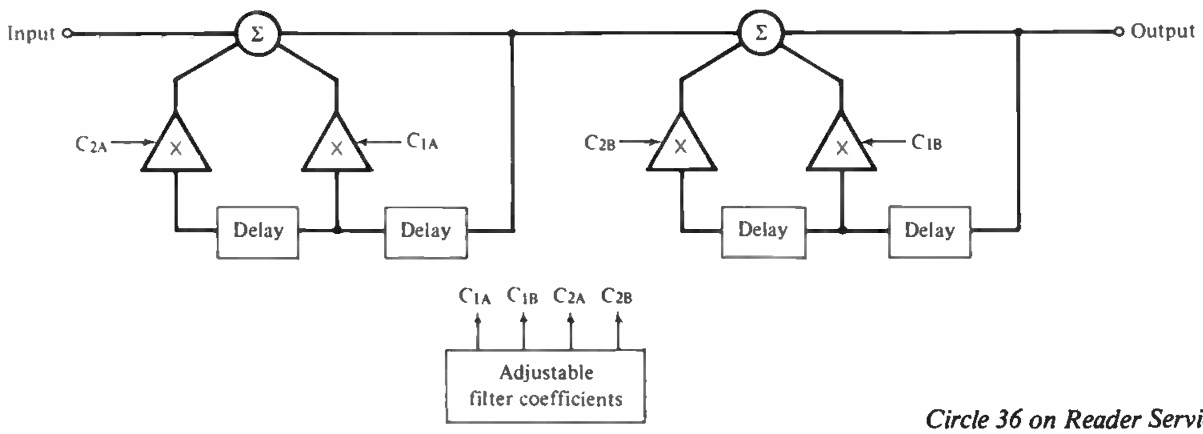
In applications where more than one filter function is needed in a short interval, it is sometimes required to manually vary the frequency characteristics with time. Such filters are being used in speech recognition experiments, where one can synthesize speech with different voice inflections by merely changing the coefficients in the filter program. To operate a digital filter in a time-variable mode, the scaling factors are made adjustable in such a way as to control the filter parameters over the desired range of frequencies and bandwidths. A second-order recursive filter of this type is depicted in FIGURE 6. The coefficients C_{1a} and C_{1b} together with C_{2a} and C_{2b} control the center frequency, while C_{2a} and C_{2b} alone affect the bandwidth, without shifting the center frequency. Using this approach, more complex filters can be assembled by increasing the number of stages, so that the controlling parameters sweep through a wider range.

ADAPTIVE FILTERS

In situations where audio signals are obscured in a noisy environment, it is possible to extract those signals from the noise in an adaptive manner, and hence optimize the signal-to-noise ratio. To accomplish this, the filter coefficients are continuously adjusted by an adaptation processor, enabling the filter to automatically track the noisy signals.

FIGURE 7 shows a commercially available adaptive digital filter system designed for application in voice intelligibility enhancement where it is necessary to wring out correlated noises, spectral distortion, acoustic resonances, reverberations, and other signal convolutions. The 5 MHz filter speed of the instrument permits the use of filter sizes of, say 445th order at 5.0 kHz and 700th order at 3.0 kHz bandwidth, with still higher orders of filtering at narrower bandwidths. Using a large transversal filter, the device estimates the noise components and subtracts them from the audio signals. The noise-reduced audio is then circulated through the system to continuously adjust the filter coefficients for maximum noise cancellation. Where undesirable periodic signals are present, such as tones, the transversal filter acts as a spectral line enhancer. For some stationary noises, such as music masking a voice, the tracking speed is adjusted for a fast convergence rate; and for less-stationary noises, such as reverberation, a

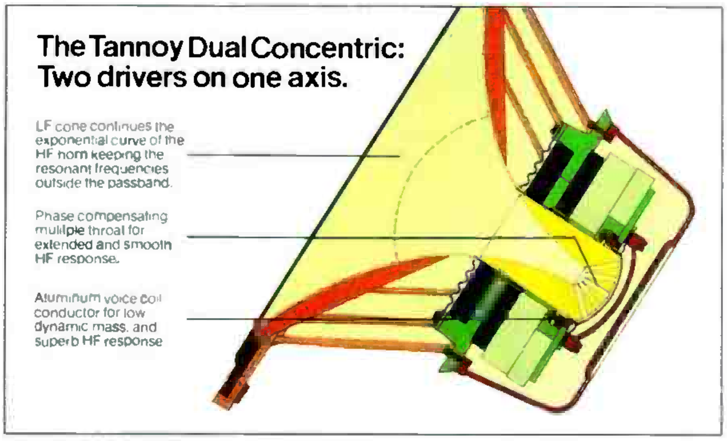
Figure 6. Cascade realization of a time-varying digital filter.



BGW SYSTEMS

TANNOY

Tannoy: A single source for the single-point sound source.



Behind the grill of most Tannoy Loudspeakers is not the usual collection of separate tweeters and bass units. Instead, you'll see what appears to be one speaker only, but which, in fact is two speakers in one.

Therein lies the Tannoy Dual Concentric principle. Single point sound. This unique method of placing a high frequency compression driver within and behind a direct radiator bass unit provides superb natural sound, low coloration and phase coherence for proper stereo imaging and high sensitivity.

A bass that's deep but not "thick." Individual bass notes should be deep and rich, but not "thick." They should be easily separated by the ear. With classical music the bass should be a fine reproduction of the natural sound of the instruments.

With contemporary rock, a loudspeaker should easily handle the accentuated bass without distortion. In Tannoy Loudspeakers, these capabilities are apparent. You can separate the drummer from the bass player and the tubas from the cellos.

Midrange: The "live" factor. This is more difficult for comparative evaluation as most loudspeakers offer acceptable qualities in this tonal range.

Choose a solo instrumental record and try not to listen to the melody which may distract, concentrate on purity of sound, a closeness to the actual instrument.

Close your eyes. Vocals should not sound "wooden" or "telephone like." Tannoy Loudspeakers provide the closeness to reality which gives that very satisfying illusion of the original "live" performance.

Clarity and response on the high end. The snap of a snare drum, the sound of an acoustic guitar, string sections, an operatic soprano, or the crash of a cymbal, these will tell you if the high frequency response is providing the absolute clarity essential for good treble quality.

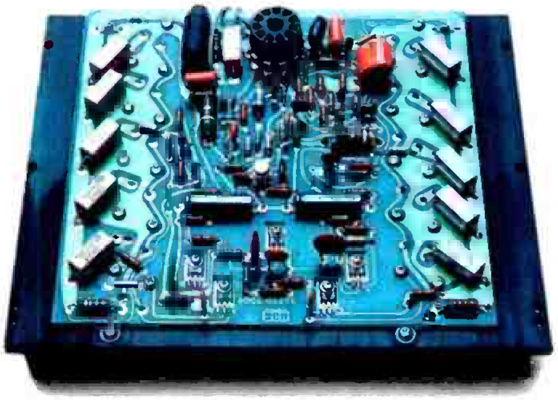
Voice sibilant should be clear but not over emphasized, the infinite flexibility of the female voice is a revealing test for any loudspeaker, and Tannoy Loudspeakers pass this test faultlessly.

The English know. Some of the best recording studios in the world are in England, and in the U.K., Tannoy is one of the most popular, if not the most professional loudspeaker.

Call 800-421-1274. We'll tell you where you can listen to a full line of Tannoy studio monitors. And discover what the English have known for over fifty years.

Model X05000 Time Compensated Electronic Crossover

BGW: Technology and engineering in a world of whistles and bells.



How do you become the world leader in power amps in five years or less? That's easy:

Make it first. Imagine. Invent. Create. You end up with things like fully modular construction, fool-proof speaker protection, and one-piece construction.

Give it more. More features. Like precision stepped thick film input attenuators, front panel mounted magnetic circuit breaker/power switch, and multi-colored, state-of-the-art LED display circuitry.

Make it last. Use welded steel instead of screws. (They unscrew.) Don't overstress the product. It'll live longer. And keep it simple. We make it modular. So servicing is that much easier.

If you like what we're saying, you'll love what you see. Test a BGW for yourself. Or, read on.

The DC Arc Interrupter: Still the best speaker protection you can buy. When harmful d.c. voltage is present at the amplifier's output, a sensing circuit triggers a fast-acting, heavy-duty relay to open and disconnect the amplifier from the loudspeaker load.

Due to the high energy associated with this type of potentially disastrous failure, conventional relay circuitry was found unacceptable and this new system, based on magnetic arc-interrupting physics was developed.

The largest SOA in the industry. Our BGW 250 series uses three times as many output devices as the competition. And since output devices are the single most important factor in determining the lifespan of an amplifier, expect ours to last three times as long.

In our 750 series, we use a heavy-duty rear-mounted two-speed industrial fan to move large volumes of air in from the back, over the internal circuitry and up through exhaust ports.

This unique thermostatic cooling system, combined with 24 large-geometry output devices, give BGW the largest SOA, or Safe Operating Area, in the industry today.

Two handles and one piece of steel. Instead of aluminum or plastic, the chassis of all BGW amps are constructed of welded steel for maximum strength, rigidity and RFI immunity.

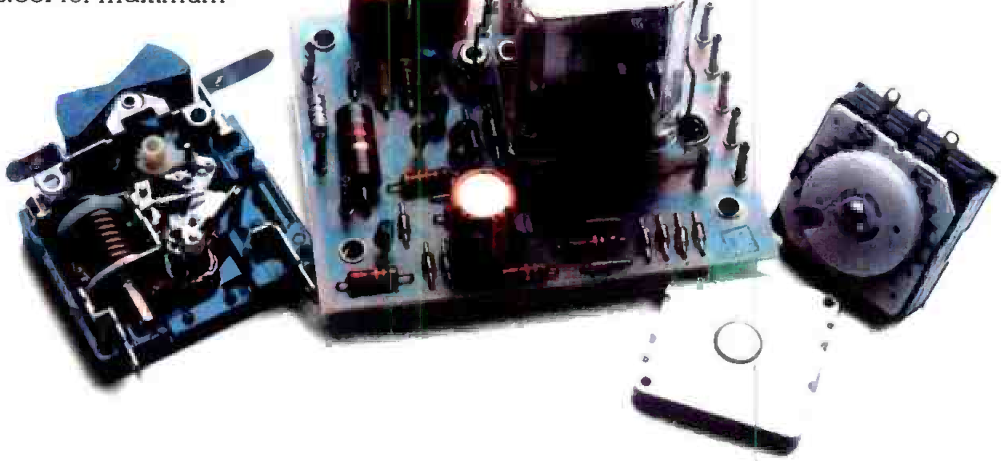
Two, large, convenient front panel mounted handles make moving easy. (By the way, because of our unusually strict standards, we are forced to make most of our metal parts ourselves. And that's fine with us.)

Circuitry: Sophistication at the heart of the system. We use large-geometry, full-complementary circuitry. Full. Not "quasi" like a lot of our competitors still use. Full costs a little more. But the result is much better high frequency response with much less distortion.

It's a tradeoff we'll gladly make.

Thanks, Billboard. Thanks, America. According to the Billboard magazine U.S. Equipment Brand Usage Survey, between 1977 and 1980, of the top three amplifier manufacturers, only BGW showed an increase in market share. Specifically, a 25% gain in studio usage, making BGW the fastest growing power amplifier source in the United States today.

That's what happens when you make it right.



Look at the Numbers:

TANNOY

MODEL	DRIVE UNIT	MAXIMUM OUTPUT LEVEL (PEAK)	SENSITIVITY 1 WATT 1 METER ANECHORC 4TT STERADIANS	FREQUENCY (1) RESPONSE	DISPERSION INCLUDED ANGLE @ –6dB POINT @ 10 KHZ	RECOMMENDED AMPLIFIER POWER @ 8 OHMS	CROSSOVER FREQUENCY	ENCLOSURE DIMENSIONS H X W X D	ENCLOSURE INTERNAL VOLUME
SRM 10B	10" Dual Concentric	109dB SPL (115dB)	90dB	55Hz-20kHz	90 degrees conical	50 Watts	1 2 kHz	20 6 X 13 8 X 10 4"	35 Liters. 1 2 Cubic Feet
SRM 12B	12" Dual Concentric	112dB SPL (117dB)	92dB	55Hz-20kHz	90 degrees conical	100 Watts	1 2kHz	23 X 15 7 X 10 8"	46 5 Liters. 1 6 Cubic Feet
M 1000	15" High Sensitivity Dual Concentric	114dB	94dB	50Hz-20kHz	90 degrees conical	200 Watts	1 0kHz	40 5 X 28 4 X 17"	230 Liters. 8 Cubic Feet
M 3000	15" Wide Bandwidth Dual Concentric	112dB SPL (119dB)	92dB	40Hz-20kHz	90 degrees conical	150-200 Watts	1kHz	40 5 X 28 4 X 17"	230 Liters. 8 Cubic Feet
DREAD-NOUGHT	1-15" Special Dual Concentric 2-15" Woofers	121dB SPL (126)	96dB	30Hz-20kHz ±3dB	90 degrees conical	750 Watts Low Frequency 500 Watts Mid Frequency 250 Watts High Frequency	250Hz 2 0kHz	35 X 52 4 X 23 2. 14 2 15° Baffle Slope	400 Liters (15 Cubic Feet) 40 Liters (1 5 Cubic Feet) Sealed Cavity

(1) Frequency Response measured in 1/3 octave bands at any power up to Rated Continuous Power with response within ± 4dB

BGW

MODEL	DESCRIPTION	TOTAL POWER* OUTPUT	8Ω POWER OUTPUT EACH CHANNEL	FEATURES MODULAR CONSTRUCTION	MAGNETIC CIRCUIT BREAKER	FULL COMPLEMENTARY CIRCUITRY	DC ARC INTERRUPTOR SPEAKER PROTECTION	FAN COOLED	CALIBRATED PRECISION STEPPED ATTENUATOR	LOSS OF FEEDBACK CLIP INDICATOR	TRI COLORED LED VU METER WITH CLIP LIGHT
75	Professional Power Amplifier	75	25 Watts	Yes	No	Yes	No	No	No	No	No
150	Professional Power Amplifier	150	50 Watts	Yes	Yes	Yes	No	No	Yes	Yes	.5% & 50% LED's
250D/E	Professional Power Amplifier	400	100 Watts	Yes	Yes	Yes	Yes	No	Yes	Yes (250D)	Yes (250E)
600	Professional Power Amplifier	800	175 Watts	Yes	No	Yes	No	No	No	Yes	No
750B/C	Professional Power Amplifier	900	225 Watts	Yes	Yes	Yes	Yes	Yes	Yes	Yes (750C)	Yes (750B)
1250	Professional Power Amplifier	1200	400 Watts	Yes	Yes	Yes	Yes	Yes	Yes	No	Yes
320B	Commercial Power Amplifier	100 Watts/Ch @ 70 volts		Yes	Yes	Yes	No	No	Yes	Yes	No
620	Commercial Power Amplifier	200 Watts/Ch @ 70 volts		Yes	Yes	Yes	No	No	Yes	Yes	No

*TOTAL POWER OUTPUT
The total power output is the actual power output as measured during our final test at the factory
Test conditions mono operation 8 ohm load 1kHz @ 0 1% Total Harmonic Distortion Line voltage maintained at 120 volts RMS 60Hz This power is equivalent to the sum of both channels when driving 4 ohm loads in the stereo mode

BGW The exclusive importer/distributor
for Tannoy products in the United States.

BGW/TANNOY

13130 S. Yukon Ave., Hawthorne, CA 90250; Telex: 66-4494, Phone: (213) 973-8090

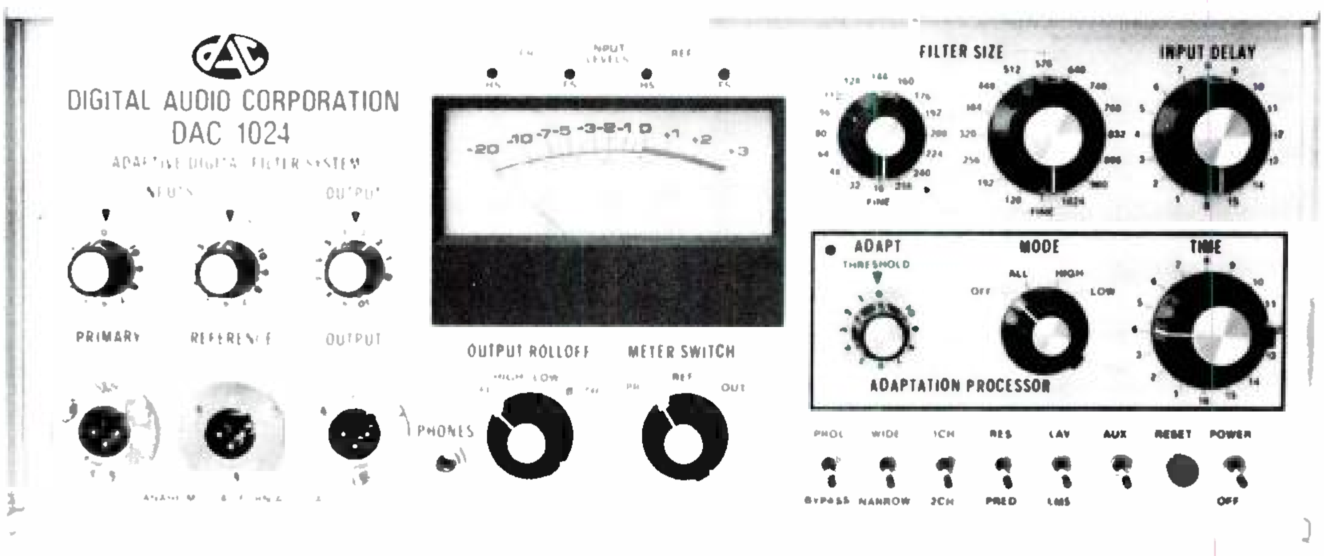


Figure 7. An adaptive digital filter system used for noise cancellation and voice intelligibility enhancement.

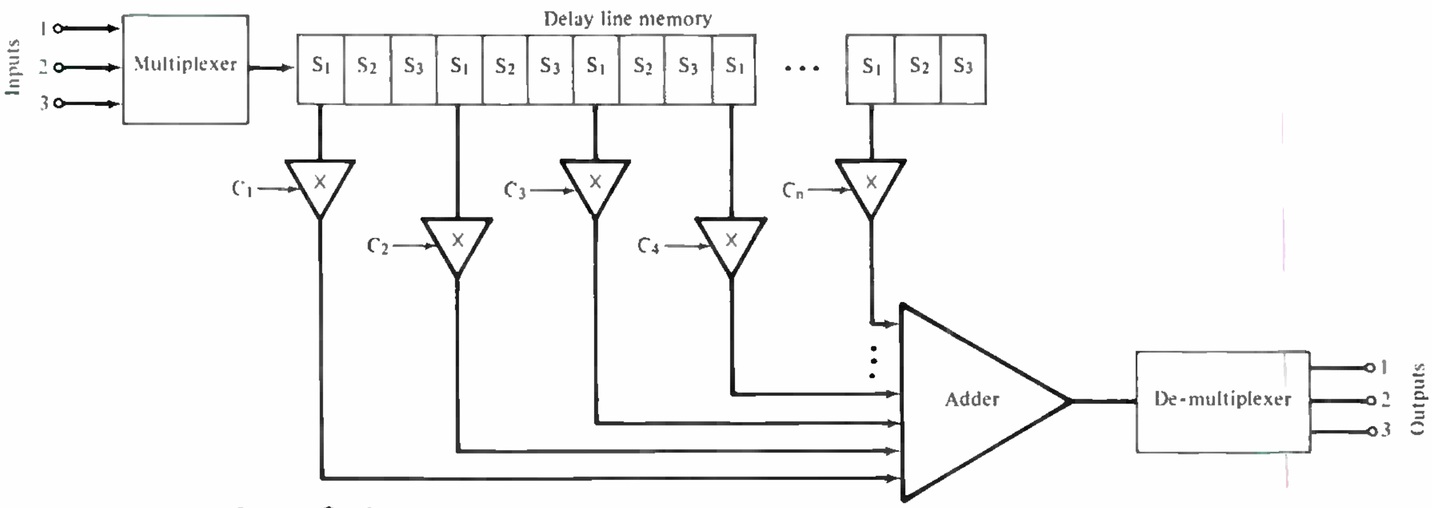
slower rate is selected. There is also a dual-channel mode of operation utilizing two separate audio inputs, one consisting of the audio signals accompanied by the noise, and the other a reference input of the noise alone. In this way, differences of phase, amplitude, and spectral shape are taken into account in the enhancement process.

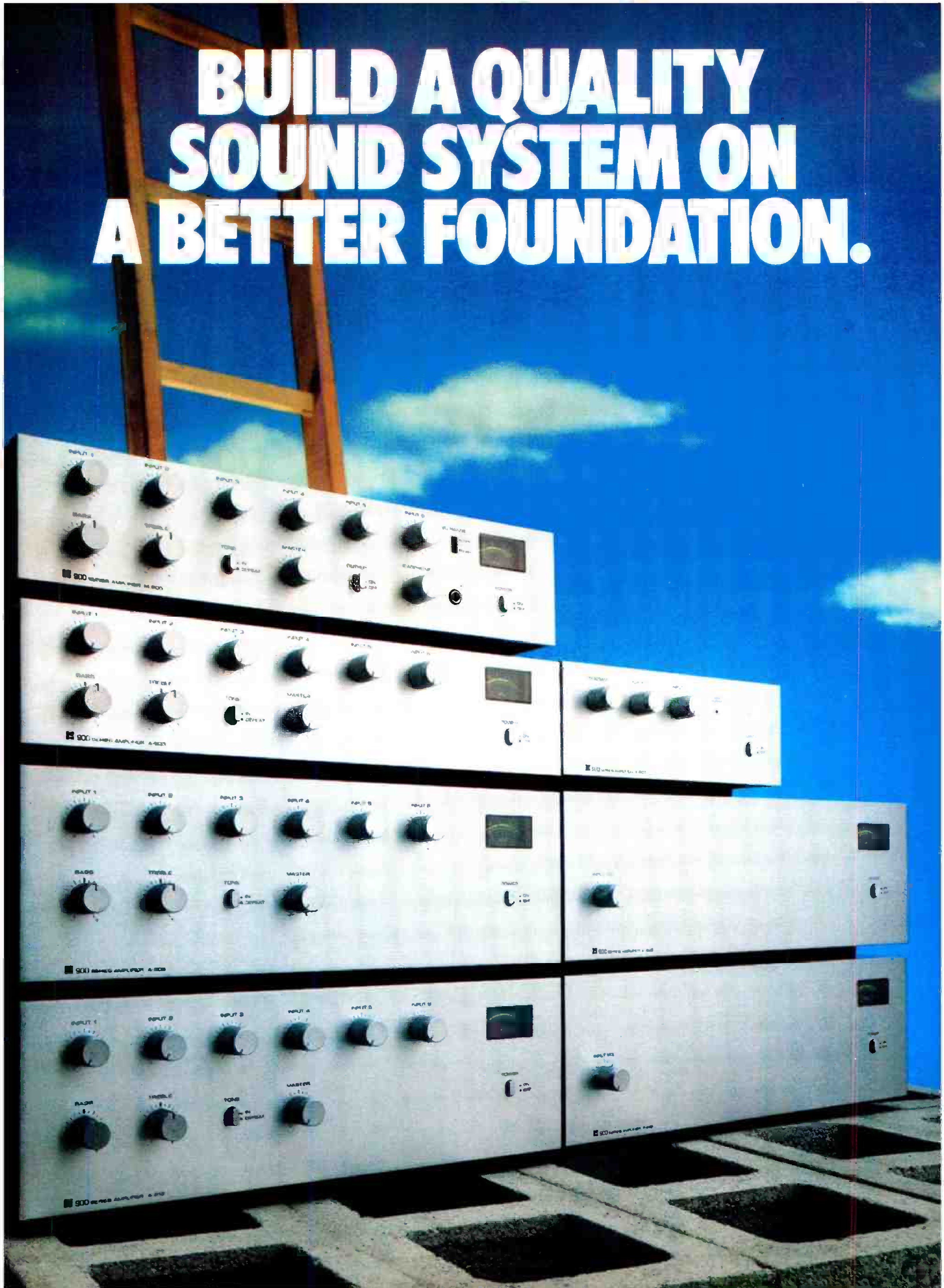
MULTIPLEXING SCHEMES

If the input data rate (sampling rate times the number of bits per sample) is well below the speed limitation of the digital circuitry, a digital filter can be multiplexed to utilize the circuitry more efficiently. Specifically, if the time intervals between samples is used to generate the next output, and if enough time exists between samples for the system logic to repeat these operations, the filter can be time-shared among numerous input signals. In FIGURE 8, for example, suppose that one filter unit is employed to equalize the inputs of three independent audio sources. Initially, the sampled input pulses are interleaved by the multiplexer, sample by sample, and applied serially to the filter input. The bit rate would be increased by a factor of three, and the shift register containing the signal samples would be three times the length of the delay line for a single input. Thus, for each channel, every third sample is effectively processed by the arithmetic unit. The arithmetic unit, in turn, performs the same multiply-add calculation as the single-input case, but operates three times faster. At the output, the digital pulse stream emerges in the same interleaved order as the input, and is thus easily separated by the de-multiplexer.

Another type of multiplexing operates on a single-input signal to effect a number of different subfilter forms. In this arrangment, the digital pulse stream is made to pass through the multiplex system more than once, to produce higher orders of filter complexity. As an example, let us consider the implementation of a 16th-order filter in cascade form using the multiplexed second-order recursive filter shown in FIGURE 9. Here, the combining of eight separate filters into one multiplexed unit requires that the bit rate in the filter be increased by a factor of eight, and the shift register delay lengths also

Figure 8. The transversal filter uses a multiplexing arrangement for time-sharing three independent inputs. Every third signal sample is processed by the filter, with S_1 representing the sample of input 1.




BUILD A QUALITY
SOUND SYSTEM ON
A BETTER FOUNDATION.

STARTING SEPTEMBER 1, THE NEW TOA 900 SERIES PA AMPLIFIERS.

LOOKS REAL EXPENSIVE, DOESN'T IT?

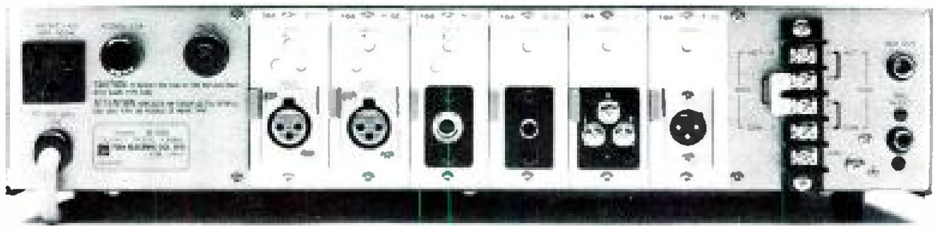
The facts of the matter are, the new TOA 900 Series is 3 to 11% less money than the same reliable 900 Series you've depended on over the years! We simply improved their performance. And, quite obviously, their looks. The new 900's are definitely affordable, just like before. And the TOA dedication to quality with the highest reliability hasn't changed either.

IF YOU DON'T KNOW ABOUT US.

You should. We've been delivering the solid-state 900's for nearly a decade. You'll find TOA PA products in churches, schools, hotels, theaters, night clubs, convention halls and auditoriums. They're working day-in, day-out in shopping centers, department stores, recreation facilities, stadiums, industrial plants and offices. For paging, announcements, background music and broadcasting, a TOA PA System is being relied upon at the very heart of almost any communications need.

THERE'S A SPECIFIC 900 FOR YOUR JOB.

Select from four MIXER/POWER amplifiers with two to six inputs and power from 10 to 120 watts RMS. There's the M900 MIXER/AMPLIFIER with six in and three models of Power Amps with 60,120 and 240 watts RMS capability. A total of eight different models for



The rear view. Simple, uncluttered. Everything you need to get the job done faster and easier.

maximum flexibility and versatility. All of our main outputs are 25 & 70 volts, transformer coupled, balanced. And some models feature transformerless direct outs. All models (except the mini 901) feature illuminated VU meters. All our mixers have full bass and treble tone equalization, too.

SO, WHAT ABOUT THE SPEC'S?

Frequency Response is 20 to 20, ±1.0 dB; THD is always less than a tenth of one percent at rated output (0.1%)!; signal-to-noise is as great as 108 dB and with gain at maximum, not less than 82 dB!; and, Output Regulation at 1 kHz is typically 1.0 dB (transformer output) and even less, 0.5 dB direct. Improved. And definitely ahead.

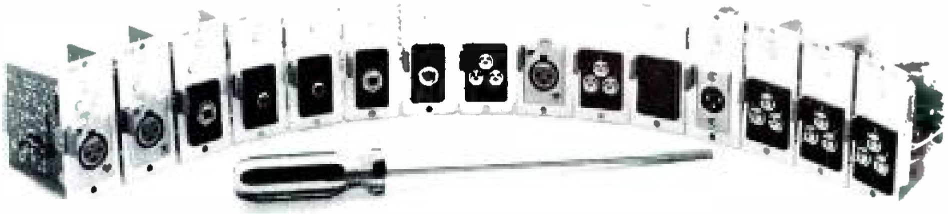
OUR BIGGEST TECHNICAL ADVANTAGES.

With a TOA you never have to worry about overload or short circuits. Because of unique circuitry, we don't require a complicated protective circuit to protect our gear from accidents and mismatched connections that can happen during your installation—

or worse yet, after you leave the scene. The line features an assortment of 15 different full-range applications plug-in modules: no wiring or fumbling for connectors. Just order XLR's, phone jacks, RCA's, 3 screw or 4 pin connectors on most everything we have from Mic Preamps to Tone Signal Generators. There's even the capability on our 6 input models to have 2 exclusive line output modules. Three of the input modules also feature optional muting. From input to output, a new 900 will get you into a tight job easier, and out faster. Much faster.

The new 900 Series—they're the proven way for you to build a quality sound system on a better foundation.

Call us at (415) 588-2538. We'll be happy to send the spec's and show you the new and improved versions of the old, reliable, and affordable PA Standards.



Choose from 15 assorted solid-state, plug-in input modules. Your only tool is a screwdriver, so pack-up your soldering iron.

Proven in the States.

TOA Sound Communications

TOA Electronics, Inc.
1023 Grandview Drive
S. San Francisco, CA 94080
(415) 588-2538
Telex: 331-332

Circle 27 on Reader Service Card

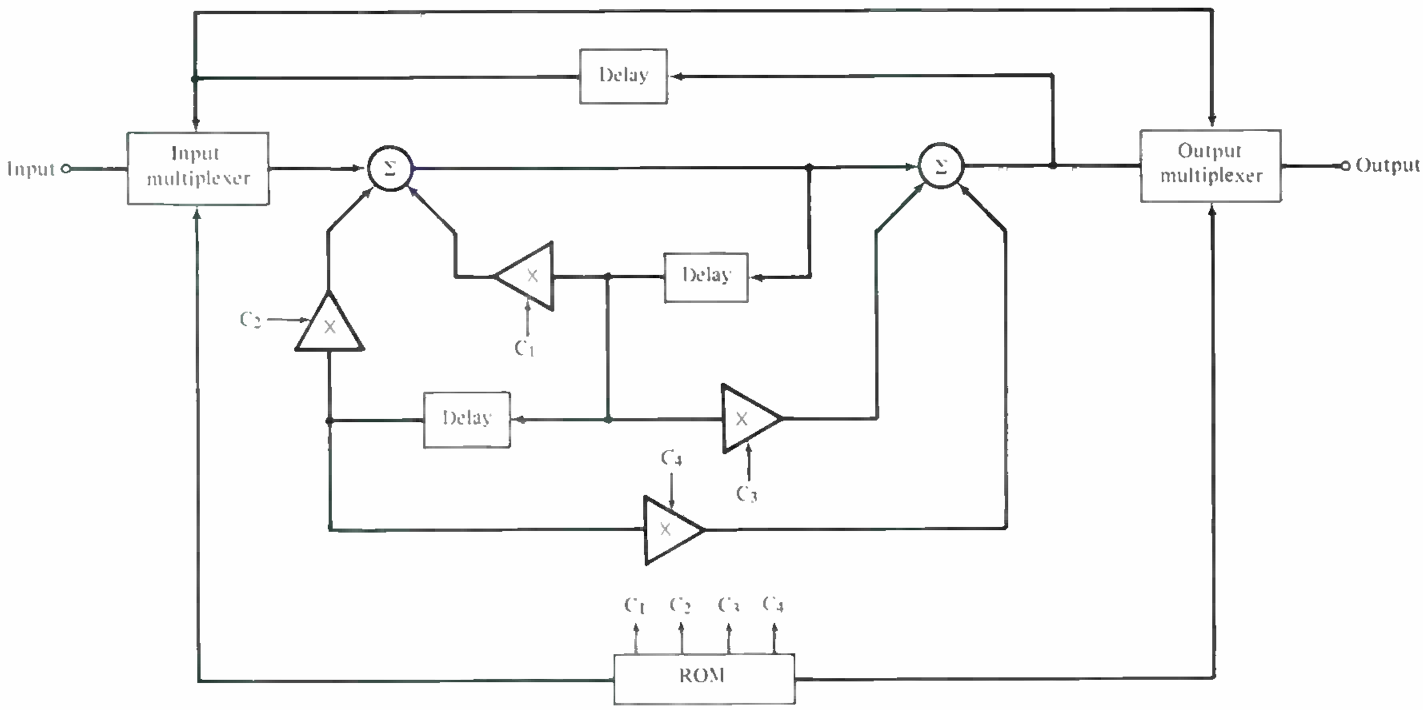


Figure 9. Multiplexing scheme where the output is repeatedly fed back to the input to effect higher orders of filtering.

be increased accordingly. The filter coefficients are supplied from the ROM which cycles the eight data values for each multiplication during every sampling interval. The data values are routed in and out of the filter by the input and output multiplexers, which are also controlled by the ROM.

During the first number of bits of each sampling interval, the input samples are processed by the arithmetic unit of the first subfilter in the cascade form. This processing takes essentially one-eighth of the basic sampling period to complete. By delaying the resulting output one-eighth of the sampling period, the data values can be fed back via the multiplexer arrangement to become the new input to the filter during the second number of bits of the sampling interval. This feedback process is repeated with the filter coefficients of the ROM changed each time to correspond to the appropriate subfilter in the cascade form. The eighth, or last, output during each sampling interval is the desired 16th-order filter output.

When low frequency filters must be realized, the requirements for coefficient precision are increased, so that it may be convenient in a multiplex operation to maintain fixed scaling factors, but operate at a reduced sampling rate. If, for example, the sampling frequency is halved, the center frequency of a bandpass filter will shift one octave lower in frequency while maintaining the same relative bandwidth. Of course, each time the sampling frequency is lowered, it must be accompanied by a low-pass filter action to correspondingly reduce the high-frequency cutoff of the sampled pulses. This procedure is necessary in order to prevent the introduction of aliasing distortion into the system. One application where this concept is utilized is the implementation of octave-band filter sets in digital frequency analyzers.

Multiplexed digital filters have found application as phase-shifting devices in modifying the tonal and spectral characteristics of electronic musical signals. To provide the filter with dynamically variable properties, the coefficients are usually stored in a small special-purpose computer, and any one of a number of filter-sweeping programs can be selected by an appropriate control function. ■

W. J. J. HOGE

Measuring Loudspeaker Driver Parameters

If you've survived Mike Math I & II, *here's some math from the other end of the signal path.*

QUITE OFTEN, a commercially built loudspeaker system will not fill a particular need in a particular situation. When this happens, it is time to design a custom system. Loudspeaker design used to be more of a black art than a science. But now, modern network synthesis techniques may be applied to the design of the loudspeaker, whether it be a direct radiator [1, 2, 3] or a horn-loaded unit [4, 5, 6].

Before attempting to design a loudspeaker system using an existing driver, one must know the values of the driver parameters which influence system performance. Unfortunately, most manufacturers still do not provide such data on their units. In this article, we will briefly review the theory of loudspeaker synthesis, examine the equivalent circuit of a loudspeaker, and the theory of driver small-signal parameter measurement. This theory will be presented in (I hope) a straightforward manner; no great math skills are required. Mixed with the theory will be some worked examples describing how to make the measurements without having to use half the H-P or B & K catalogs worth of equipment. So get out your snow shovels (or manure forks), and here we go.

A REVIEW OF SOME THEORY

Novak [7] presented an equivalent circuit for a loudspeaker, which Thiele [1] uses as the basis for the application of modern network synthesis techniques to direct-radiator loudspeakers. FIGURE 1 gives a general low-frequency acoustical equivalent circuit of a direct-radiator loudspeaker. In the figure, each electrical circuit element represents a specific acoustical property of the loudspeaker and its enclosure.

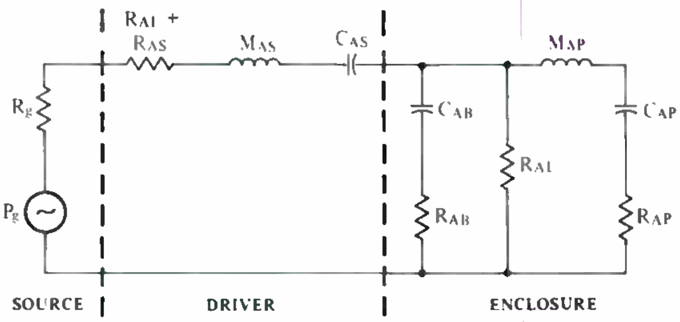


Figure 1. The acoustical equivalent circuit of a direct-radiator loudspeaker.

In FIGURE 1, the circuit elements are defined as follows:

- P_g equivalent pressure source of the electrical energy source,
- R_g acoustic equivalent source output resistance,
- R_{AE} acoustic equivalent of the voice coil resistance,
- R_{AS} acoustic resistance of the driver suspension losses,
- M_{AS} acoustic mass of the driver diaphragm including air load,
- C_{AS} acoustic compliance of the driver suspension,
- C_{AB} acoustic compliance of the air in the enclosure,
- R_{AB} acoustic resistance caused by the enclosure's absorption losses,
- R_{AL} acoustic resistance caused by enclosure leaks,
- M_{AP} acoustic mass of the passive radiator port,
- C_{AP} acoustic compliance of the passive radiator suspension, and
- R_{AP} acoustic resistance of the port or passive radiator losses.

W. J. J. Hoge is a research and development engineer for Harrison Systems, Nashville, TN.

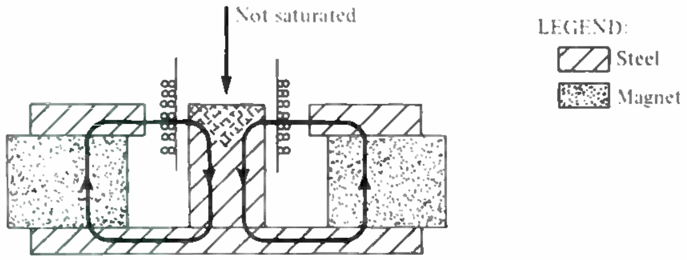


Figure 2. Section view of a loudspeaker motor.

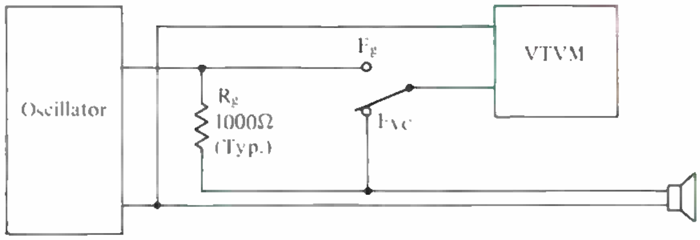


Figure 3. A simple test for driver resonance.

Thiele noted that the circuit resembles a fourth-order, high-pass filter, and he derived relationships among the circuit elements for several different response shapes. His Table 1 allows us to design a system by specifying the ratios between: the system's half-power (−3 dB) frequency and the driver's resonance frequency (f_3/f_S), the half-power frequency and the enclosure's resonance frequency (f_3/f_B), and the acoustic compliance of the driver suspension and the compliance of the air in the enclosure (f_{AS}/f_{AB}). An additional item, the total quality factor of the driver (Q_{TS}) must also be specified.

The compliance, C_{AB}, is a function of the volume of the enclosure and the effective area of the driver diaphragm. Therefore,

$$C_{AS}/C_{AB} = V_{AS}/V_B \tag{1}$$

where V_{AS} the volume of air which has the same compliance as the driver suspension, and
V_B the volume of the enclosure.

The other low-frequency small-signal performance specification is efficiency. The half-space efficiency of a driver is given by

$$\eta_o = \frac{4\pi^2}{c^3} \cdot \frac{f_{SA}{}^3 V_{AS}}{Q_{ES}} \tag{2}$$

where η_o the driver's half-space efficiency,
c the speed of sound, in meters-per-second, and
V_{ES} the quality factor of the driver, considering electrical losses only.

To determine the total quality factor of the driver, we must consider mechanical losses as well. Thus,

$$Q_{TS} = \frac{Q_{ES}Q_{MS}}{Q_{ES} + Q_{MS}} \tag{3}$$

where Q_{TS} the total quality factor of the driver, and
Q_{MS} the quality factor of the driver, considering mechanical losses only.

At high frequencies, the voice coil inductance begins to effect system performance. However, voice coil inductance is not constant with frequency. Consider the loudspeaker motor of Figure 2. The voice coil rides around the center core. Part of the core is saturated with flux from the magnetic field, but part of it is not. As the voice coil moves in and out, a greater or lesser number of turns in the coil surround the unsaturated section, and the inductance of the coil varies. In some drivers, variations of +100 −50 percent are not uncommon. The voice coil's inductance and resistance cause the high frequency response to roll off (they amount to a single-pole filter). The corner frequency varies with the large-signal inductance variation and this causes amplitude modulation of the higher frequencies. The effective small-signal inductance, L_E, gives an indication of the frequency region where this problem may occur. It may also be used to estimate interaction of the driver and the crossover. The voice coil resistance, R_E, is also necessary before beginning to design the crossover.

So far, we have found that the following small-signal driver parameters are required in order to design a loudspeaker:

- f_{SA} driver resonance frequency in free air,
- V_{AS} compliance equivalent volume,
- Q_{ES} driver quality factor considering electrical losses only (2),
- Q_{MS} driver quality factor considering mechanical losses only,
- Q_{TS} total driver quality factor (3),
- L_E effective voice coil inductance, and
- R_E voice coil resistance.

How can we measure them?

The measurement of L_E and R_E is a trivial matter. A bridge capable of good resolution and accuracy can be used to read these data directly from the driver's input terminals. Likewise, the determination of f_S is not difficult. An oscillator, voltmeter, and resistor can be used, as in Figure 3. The voltmeter reading will peak at the resonance frequency.

V_{AS} may be derived from the mechanical compliance of the suspension, C_{MS}. We may attempt to measure this directly by placing the driver on its back (with the cone facing upward and the cone axis vertical) and putting a known mass on the cone. Although the measured displacement of the cone might be used to compute the compliance, this technique ignores the effect of the mass of the moving system. This mass causes a downward force on the suspension which biases it off center, since the suspension of a loudspeaker is not necessarily linear. This is particularly true in high-compliance drivers, which generally have high-mass cones. In these cases, the cone mass results in a biasing force which can drive the suspension far enough off-center to destroy the value of the measurement. There must be a better way.

Consider the low-frequency electrical equivalent circuits of a driver in free air, and of a closed-box loudspeaker with a lossless enclosure, as shown in FIGURE 4.

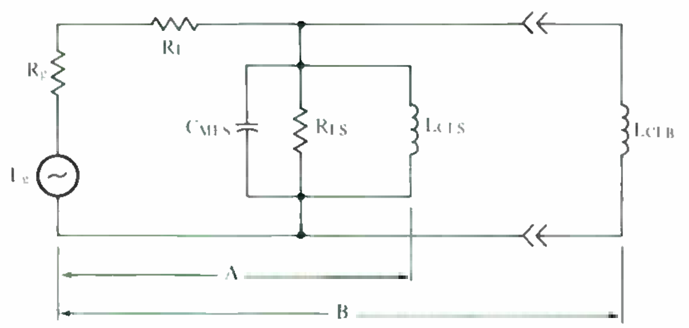


Figure 4. Electrical equivalent of a driver in free air (A). The addition of L_{CEB} makes the circuit the equivalent of a closed-box speaker in a lossless enclosure (B).

The circuit elements in FIGURE 4 are defined as follows:

- E_g open circuit output potential of the source,
- R_g source output resistance,
- R_1 voice coil resistance,
- C_{MES} electrical capacitance caused by the driver mass including air loads,
- R_{ES} electrical resistance caused by the driver suspension losses,
- L_{CES} electrical inductance caused by the driver suspension compliance, and
- L_{CEB} electrical inductance caused by the compliance of the air in the enclosure.

We can determine the value of L_{CES} and thus of V_{AS} by measuring the resonance frequency of the driver in free air and then on a lossless test box and performing some calculations. At first glance it may seem that the lower effective inductance is the only difference between the two circuits. However, this is not true. C_{MES} is the capacitance resulting from the moving mass including air loads. As Beranek [8] points out, one effect of the test box is to increase the air load above that in free air. We must account for this variation. Let us define

- C_{MES1} electrical capacitance in free air, and
- C_{MES2} electrical capacitance mounted on an enclosure.

Now the two resonance frequencies are given by

$$f_{SA} = \frac{1}{2\pi(C_{MES1}\, L_{CES})^{1/2}} \tag{4}$$

and

$$f_{CT} = \frac{1}{2\pi\left[C_{MES2}\left(\dfrac{L_{CEB}\, L_{CES}}{L_{CEB} + L_{CES}}\right)\right]} \tag{5}$$

where f_{SA} is the resonance frequency of the driver in free air and f_{CT} is the resonance frequency of the driver mounted on the test box. Equations (4) and (5) may be rewritten in the forms

$$f_{SA} = \frac{1}{2\pi\,[C_{AS}\,(M_{AD} + 2M_{A1})]^{1/2}} \tag{6}$$

and

$$f_{CT} = \frac{1}{2\pi\left[\dfrac{C_{AB}\, C_{AS}}{C_{AB} + C_{AS}}\,(M_{AD} + M_{A1} + M_{AB})\right]^{1/2}} \tag{7}$$

where M_{AD} is the acoustic mass of the diaphragm and voice coil, M_{A1} is the acoustic mass of the air load on the front of the diaphragm, and M_{AB} is the acoustic mass of the air load on the back of the diaphragm when mounted on the enclosure. From Beranek,

$$M_{A1} = \frac{0.23}{a} \tag{8}$$

and

$$M_{AB} = \frac{B_B \rho_0}{\pi a} \tag{9}$$

where a is the effective radius of the diaphragm, ρ_0 is the density of air, and B_B is "Beranek's B" (a fudge factor dependent on the ratio of the effective area of the diaphragm to the area of the side of the enclosure on which it is mounted).

We may now calculate C_{AS} by rewriting equation (6) or (7). We may then find V_{AS} by

$$V_{AS} = \rho_0 c^2 C_{AS} \tag{10}$$

The quality factors Q_{TS}, Q_{ES} and Q_{MS} may be determined from terminal impedance data taken on the driver in free air. Consider FIGURE 5, a plot of the magnitude of the voice coil impedance as a function of frequency. The resonance frequency, f_{SA}, is located where the impedance is maximum. The ratio of the magnitude of the voice coil impedance $|Z_{VC}|$ at f_S to the DC resistance R_E is defined as

$$r_0 = \frac{|Z_{VC}|_{max}}{R_E} \tag{11}$$

The two frequencies f_1 and f_2 are found where $|Z'_{VC}| = R_E\sqrt{r_0}$. Then, from circuit theory we know that

$$Q_{MS} = \frac{f_{SA}\sqrt{r_0}}{f_2 - f_1} \tag{12}$$

$$Q_{ES} = \frac{Q_{MS}}{r_0 - 1} \tag{13}$$

and

$$Q_{TS} = \frac{f_{SA}}{(f_2 - f_1)\sqrt{r_0}} \tag{14}$$

We now know the sort of raw data which we need and how it should be processed to yield the proper parameters. Let us examine several methods of taking the data and calculating the results.

TEST METHODS

The first method we will examine is suitable for use with limited equipment or for quick checks of the appropriate driver parameters. After measuring R_1 and L_E, the driver is connected to a test set such as shown in FIGURE 3. With the oscillator output adjusted for a maximum output current on the order of 10 mA, the resonance frequency of the driver (f_{SA}) is measured. (Note: The dial calibration of most oscillators is not sufficiently good at low frequencies. A counter may be useful.) Compute $|Z_{VC}|_{max}$ by

$$|Z_{VC}|_{max} = \frac{E_{VC}\, R_g}{E_g - E_{VC}} \tag{15}$$

where E_{VC} is the voice coil potential, E_g is the source output potential, and R_g is the series resistance between the source and the driver. r_0 is given by equation (11). Next find the two frequencies, f_1 and f_2, where $|Z_{VC}| = R_E\sqrt{r_0}$. Equations (12), (13), and (14) give the values of Q_{MS}, Q_{ES}, and Q_{TS}.

The driver is now mounted on a test box. Remember the test box is supposed to be lossless. The only aperture in the test box should be the one to which the driver is mounted, and the driver should be tightly gasketed. Any leaks may cause trouble with the measurement. This includes leaks in the driver caused by vented or porous dust caps. Also, the box must not contain any damping material which would cause absorption losses. Measure the new resonance frequency f_{CT} and $|Z'_{VC}|$ max. Now,

find the two frequencies f_1' ($< f_{CT}$) and f_2' ($> f_{CT}$) where $|Z'_{VC}| = R_E\sqrt{r_C}$. ($r_C = |Z'_{VC}|_{max}/R_E$. Small [2] defines the electrical quality factor for f_{CT} as

$$Q_{ECT} = \frac{F_{CT}\sqrt{r_C}}{(f_2' - f_1')\,(r_C - 1)} \tag{16}$$

By doing some algebra on several of the previous equations we may show that

$$V_{AS} = V_B\left(\frac{f_{CT}Q_{ECT}}{f_S Q_{ES}} - 1\right) \tag{17}$$

Worked example 1: Given the following instrument readings, find Q_{MS}, Q_{ES}, Q_{TS}, V_{AS} and η_0.

Frequency	Counter (Hz)	Voltmeter (V)	Measured in
f_{SA}	35	0.29 $\lvert E_{VC}\rvert$ max	free air
f_1	21	0.13	free air
f_2	59	0.13	free air
f_{CT}	79	0.28 $\lvert E'_{VC}\rvert$ max	lossless test box
f_1'	62	0.13	lossless test box
f_2'	100	0.13	lossless test box

R_E is 5.6Ω, R_g is 1000Ω, and E_g is 10V in all cases, and V_B is 0.040 m^3.

From equation (15),

$$|Z_{VC}|\max = \frac{(0.29)\,(1000)}{(10 - 0.29)} = 30\Omega.$$

From equation (11),

$r_0 = 30/5.6 = 5.4$, and therefore,

$|Z_{VC}| = R_E\sqrt{r_0} = 30\sqrt{5.4} = 13\Omega.$

Thus, $|E_{VC}|$ at f_1 and f_2 should be 0.13V, which checks with the data. From equations (12), (13) and (14),

$$Q_{MS} = \frac{(35)\,(5.4)^{1/2}}{(59 - 21)} = 2.1$$

$$Q_{ES} = \frac{2.1}{(5.4 - 1)} = 0.48$$

$$Q_{TS} = \frac{35}{(59 - 21)\,(5.4)^{1/2}} = 0.40$$

Again, using equations (15) and (11),

$$|Z'_{VC}|\max = \frac{(0.28)\,(1000)}{(10 - 0.28)} = 29\Omega$$

$$r_C = 29/5.6 = 5.2\mu$$

$$|Z'_{VC}| = R_E\sqrt{r_C} = 5.6\sqrt{5.2} = 13\Omega.$$

The fact that $|Z'_{VC}|$ max is somewhat lower than $|Z_{VC}|$ max indicates a box loss; however, this magnitude of difference is typical of this measurement method. Remember, the answers are approximations.

Using equation (16), we find that

$$Q_{ECT} = \frac{(79)\,(5.2)^{1/2}}{(100 - 62)\,(5.2 - 1)} = 1.1.$$

From equation (17) we find that

$$Q_{AS} = 0.040\left[\frac{(79)\,(1.1)}{(35)\,(0.48)} - 1\right] = 0.17\ \text{m}^3.$$

Finally, equation (2) may be used to find

$$\eta_0 = \frac{4\pi^2}{(345)^3} \cdot \frac{(35)^3\,(0.17)}{(0.48)}$$

Thiele suggests an improved method for making these measurements if it becomes impossible to seal the loudspeaker to the test box. A vent is added to the box. Now the magnitude of the terminal impedance plot as a function of frequency looks like FIGURE 7. Using the test set of FIGURE 3 we find; the lower impedance peak frequency (f_L), the upper impedance peak frequency (f_H), and the resonance frequency of the enclosure (f_B).

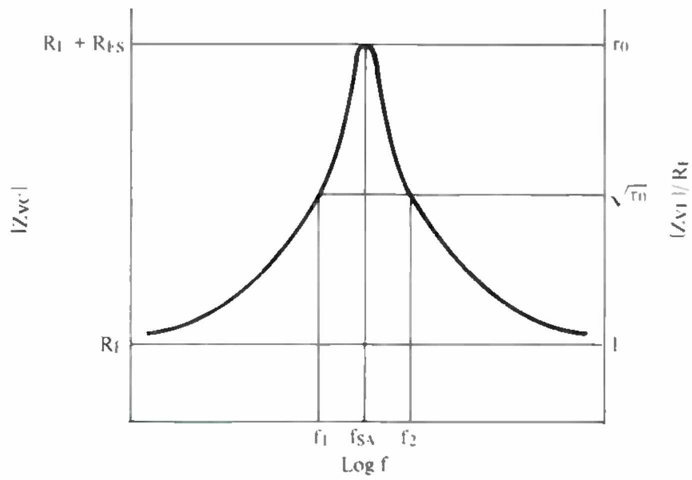


Figure 5. Driver voice coil impedance magnitude, as a function of frequency (lossless closed box).

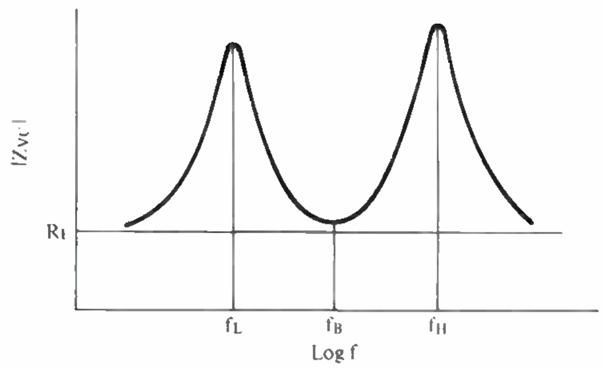
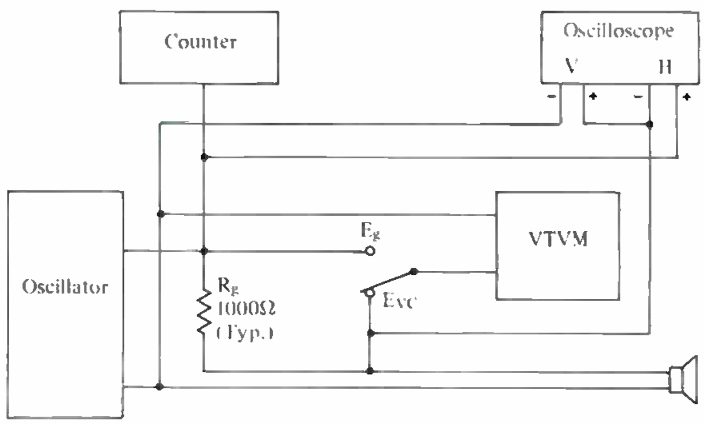


Figure 6. Magnitude of voice coil terminal impedance as a function of frequency (vented box).

Figure 7. An improved test setup.



Special care is needed in reading f_B because the curve has a flat bottom. An oscilloscope may be added, as shown in FIGURE 7. Lissajous figures will make f_L, f_H, and f_B easier to detect as the phase angle goes through zero degrees at each of these points.

From these readings we may find the value of the resonance frequency of the driver, f_{SB}, with the added air mass caused by being mounted on the test box.

$$f_{SB} = \frac{f_H f_L}{f_B} \quad (18)$$

We may find V_{AS} by

$$V_{AS} = \frac{V_B[(f_H^2 - f_B^2)(f_B^2 - f_L^2)]}{f_H^2 f_L^2} \quad (19)$$

Worked Example 2: The driver tested in example 1 is now rechecked using a vented test box. Given the following instrument readings, find Q_{MS}, Q_{ES}, Q_{TS}, V_{AS}, and η_0.

Frequency	Counter (Hz)	Voltmeter (V)	Measured in
f_{SA}	35	0.29	free air
f_1	21	0.13	free air
f_2	59	0.13	free air
f_L	13	—	vented test box
f_B	33	—	vented test box
f_H	81	—	vented test box

R_E is 5.6Ω, R_g is 1000Ω, and E_g is 10 V in all cases. V_B is 0.040 m^3. From the methods used in example 1, we find

$Q_{MS} = 2.1$
$Q_{ES} = 0.48$
$Q_{TS} = 0.40.$

Using equation (19),

$$V_{AS} = \frac{0.040[(81)^2 - (33)^2][(33)^2 - (13)^2]}{(81)^2 (13)^2}$$

$V_{AS} = 0.18$ m^3.

The reference efficiency is found with equation (2). However, we will use f_{SB} rather than f_{SA}. This is because the air load when operating on the test box more closely represents actual conditions in normal use than does the load in free air. Thus, from equation (18),

$$f_{SB} = \frac{(81)(13)}{33} = 32 \text{ Hz}$$

$$\eta_0 = \frac{4\pi^2}{(345)^3} \quad \frac{(32)^3 (0.18)}{0.48}$$

$\eta_0 = 0.012$ (or 1.2 percent).

In both examples the calculation of Q_{MS}, Q_{ES}, and Q_{TS} depends on a single set of data. Ashley and Swan [9] suggest a method which uses both magnitude and phase angle data on the terminal impedance at f_1 ($<f_{SA}$) and f_2 ($>f_{SA}$) to compute the values of the mechanical resistance of the suspension R_{MS} and the electromagnetic coupling coefficient, Bl. The values computed for the two sets of data are averaged and the mean values used to compute the quality factors. Ashley and Swan use a test set not unlike FIGURE 7; the oscilloscope is used to make the phase angle measurements. The author prefers to use a vector voltmeter or a gain/phase meter. Minor errors in reading the Lissajous figure on the oscilloscope can cause 5 percent (or more) error in the final computations. R_{MS} and Bl are found by

$$R_{MS} = \frac{[1 - (f_X/f_{SA})^2](Z_X\cos\theta_X - R_E)}{2\pi f_X C_{MS} Z_X \sin\theta_X} \quad (20)$$

$$Bl = \left\{ \frac{[(f_X/f_{SA})^2 - 1][Y_X R_E \cos\theta_X - 1]}{2 f_X C_{MS} Y_X \sin\theta_X} - R_{MS} R_E \right\}^{1/2} \quad (21)$$

where f_X is the test frequency, Y_X is the magnitude of the voice coil terminal admittance at f_X, Z_X is the magnitude of the voice coil terminal impedance at f_X, and θ_X is the phase angle of the voice coil terminal impedance at f_X.

$$C_{MS} = \frac{V_{AS}}{\rho_0 c^2 S_D^2} \quad (22)$$

We may compute the quality factors

$$Q_{MS} = \frac{1}{2\pi f_S C_{MS} R_{MS}} \quad (23)$$

and

$$Q_{ES} = \frac{2\pi f_S M_{MS} E_E}{(Bl)^2} \quad (24)$$

Q_{TS} may be found using equation (3).

Ashley and Swan also use a magnitude and phase angle reading of the terminal impedance at a frequency above the second resonance in free air to determine the effective voice coil inductance L_E. At high frequencies where the motional impedance is trivial, we can model the terminal impedance as a series connection of L_E and R_E.

$$L_E = \frac{Z_X \sin\theta_X}{2\pi x}$$

LARGE SIGNAL PARAMETERS

So far, we have ignored large-signal parameters which relate to power handling and distortion. This is because little work has been done to measure parameters such as the maximum diaphragm displacement, X_{max}. Ashley proposes measuring the maximum potential which can be applied at f_{SA} and still maintain linear operation, and, from this, calculating X_{max}. He determines the limit by observing non-linearities in the Lissajous figure on the oscilloscope. This method, in the author's opinion, depends too much on the operator's judgment. However, the author has been using it for approximate results. This is an area where more work is needed.

SOURCES OF ERROR

In several years of manual data taking, the largest single source of error in these tests was boredom-induced operator error. The errors included reading the instruments improperly and poorly sealing the driver to the test box. Other errors resulted from neglecting the effects of such things as atmospheric pressure. This was discovered when data taken in Colorado (altitude 1800 m) and Tennessee (altitude 180 m) on the same driver were radically different. Attention to detail is required.

CONCLUSION

Driver parameter information is important to a systems designer. Using the simple, though tedious, methods outlined above, one can find the parameters of the drivers he must work with. This is an important step in designing a new system or analyzing an existing one. ■

REFERENCES

1. A. N. Thiele, "Loudspeakers in Vented Boxes," *Proc. IRE (Australia)*, vol. 22, pp. 487-508 (Aug. 1961). [reprinted in *Jour. of The Audio Engineering Society*, vol. 19, no. 5, 6 (May, June 1971)]
2. R. H. Small, *Direct-Radiator Electrodynamic Loudspeaker Systems*, Ph. D Thesis, Univ. of Sydney, 1972.
3. J. E. Benson, "Theory and Design of Loudspeaker Enclosures: Part 3—Introduction to Synthesis of Vented Systems," *AWA Tech. Rev.*, vol 14, pp. 369-484.
4. R. H. Small, "Suitability of Low-Frequency Drivers for Horn-Loaded Loudspeaker Systems," AES Pre-print No. 1251.
5. D. B. Keele, Jr., "Low-Frequency Horn Design Using Thiele/Small Driver Parameters," AES Pre-print No. 1250.
6. W. J. J. Hoge, "State-Variable Analysis of Horn Loudspeakers," presented at the 57th Convention of the Audio Engineering Society, Los Angeles, 1977.
7. J. F. Novak, "Performance of Enclosures for Low-Resonance High-Compliance Loudspeakers," *IRE Trans. Audio*, vol. AU-7, pp. 5-13 (Jan.-Feb. 1959).
8. L. L. Beranek, *Acoustics*, McGraw-Hill, New York, 1954.
9. J. R. Ashley and M. D. Swan, "Improved Measurement of Loudspeaker Parameters," AES Pre-print No. 803.

db Application Notes

Computing Loudspeaker Parameters

THE BASIC COMPUTER program printed here will quickly calculate the values found in the author's worked examples 1 and 2.

Lines 30 through 200 allow the operator to input the various known values. The TAB instructions merely organize the inputted values to more-or-less conform to the layout within the article. From line 210 on, each three-line sequence of instructions calculates a value, prints its identifying symbol (or a reasonable facsimile) and then prints the value just calculated.

In Hoge's worked example 2, E'_{VC} is not used. Therefore, at line 150, the operator would input a zero. Now, at line 390, the program branches to line 590, skipping the unnecessary calculations in between. V_{AS} (=VA) is now calculated in line 590 (Hoge's equation 19) instead of in line 520 (equation 17). Next, F_{SB} is calculated and printed, and the program returns to line 530 to print V_{AS} and finally, calculate the driver's half-space efficiency in line 550, where the value of 9.6139E-7 merely represents $4\pi^2/c^3$, as required in equation 2.

Wherever possible, the program variables closely match the author's style. Thus, in line 190, the original E_G is printed out as E(G). However, to keep the use of parentheses within reason (just look at 490 or 590!), E(G) becomes EG in the actual calculating instructions.

```
 1Ø  HOME
 2Ø  DIM F(22)
 3Ø  INPUT "F(SA) = ";FS
 4Ø  INPUT "F(1)  = ";F(1)
 5Ø  INPUT "F(2)  = ";F(2)
 6Ø  PRINT
 7Ø  INPUT "F(CT) = ";FC
 8Ø  INPUT "F'(1) = ";F(11)
 9Ø  INPUT "F'(2) = ";F(22)
1ØØ  VTAB 1
11Ø  HTAB 2Ø
12Ø  INPUT "E(VC) = ";E1
13Ø  VTAB 5
14Ø  HTAB 2Ø
15Ø  INPUT "E'(VC)= ";E2
16Ø  VTAB 1Ø
17Ø  INPUT "R(E)  = ";RE
18Ø  INPUT "R(G)  = ";RG
19Ø  INPUT "E(G)  = ";EG
2ØØ  INPUT "V(B)  = ";VB
21Ø  ZX = E1*RG/(EG - E1)
22Ø  PRINT "Z(VC) = ";
23Ø  PRINT TAB(15)ZX
24Ø  RO = ZX/RE
25Ø  PRINT "RO    = ";
26Ø  PRINT TAB(15)RO
27Ø  XZ = RE*SQR(RO)
28Ø  PRINT "Z(F)  = ";
29Ø  PRINT TAB(15)XZ
3ØØ  QM = FS*SQR(RO)/(F(2) - F(1))
31Ø  PRINT "Q(MS) = ";
32Ø  PRINT TAB(15)QM
33Ø  QE = QM/(RO - 1)
34Ø  PRINT "Q(ES) = ";
35Ø  PRINT TAB(15)QE
36Ø  QT = FS/((F(2) - F(1))*SQR(RO))
37Ø  PRINT "Q(TS) = ";
38Ø  PRINT TAB(15)QT
39Ø  IF E2 = Ø THEN 59Ø
4ØØ  ZY = E2*RG/(EG - E2)
41Ø  PRINT "Z'(VC)= ";
42Ø  PRINT TAB(15)ZY
43Ø  RC = ZY/RE
44Ø  PRINT "RC    = ";
45Ø  PRINT TAB(15)RC
46Ø  YZ = RE*SQR(RC)
47Ø  PRINT "Z'(F) = ";
48Ø  PRINT TAB(15)YZ
49Ø  QC = FC*SQR(RC)/((F(22) - F(11))*(RC - 1))
5ØØ  PRINT "Q(ECT)= ";
51Ø  PRINT TAB(15)QC
52Ø  VA = VB*(FC*QC/(FS*QE) - 1)

53Ø  PRINT "V(AS) = ";
54Ø  PRINT TAB(15)VA
55Ø  H = 9.6139E-7*FS^3*VA/QE
56Ø  PRINT "H     = ";
57Ø  PRINT TAB(15)H
58Ø  END

59Ø  VA = VB*(F(22)^2 - F(11)^2)*(F(11)^2 - FC^2)/(F(22)^2*FC^2)
6ØØ  FS = F(22)*FC/F(11)
61Ø  PRINT "F(SB) = ";
62Ø  PRINT TAB(15)FS
63Ø  GOTO 53Ø
```

Taken for granted.

In most studios, the Studer B67 gets considerable use. But not much attention. It merely does everything it's supposed to do, with a minimum of fuss and bother.

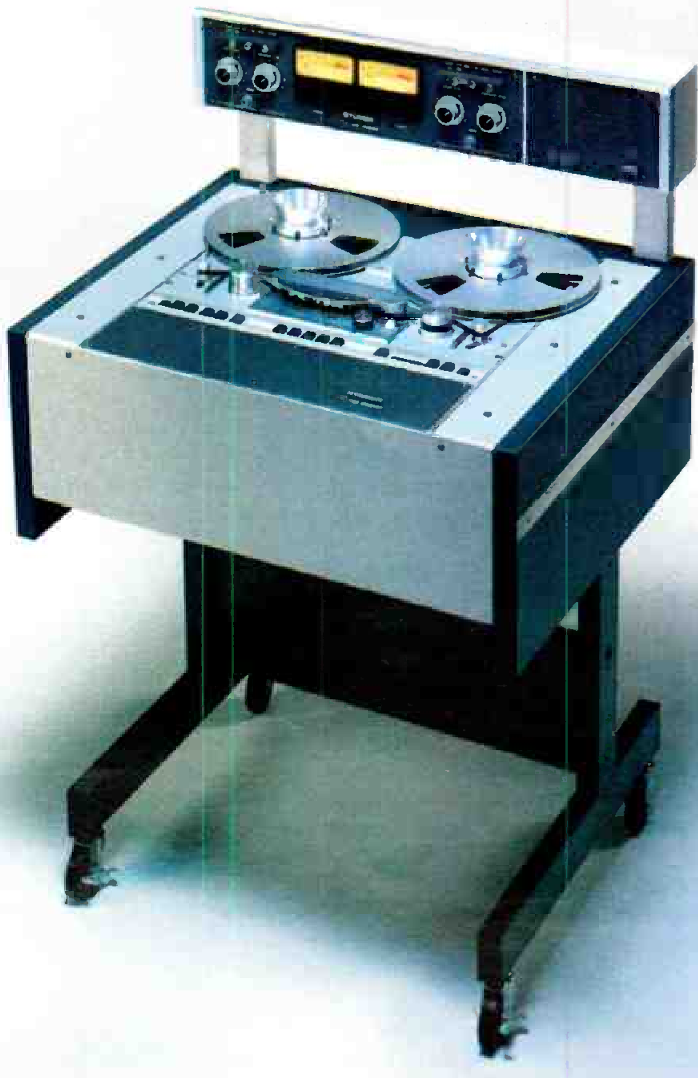
True, most engineers appreciate the smooth performance and thoughtful features. The quartz PLL capstan drive and servo-controlled spooling motors. The three speed (15/7.5/3.75 or 30/15/7.5 ips) operation. Headblocks that plug in with no need for realignment. Fader start, dump edit, and a real time counter with plus or minus readout. Improvements on the new B67 MKII include locking tension sensor arms and better head access for easier editing, plus record drop-out by punching the "PLAY" button.

In the end, though, engineers mainly applaud the way a B67 does what they expect it to do... without doing the unexpected. Year after year.

In this uncertain business, it's good to have something you can simply take for granted. Studer quality. Contact us directly for details on the affordably priced B67 MKII.

STUDER REVOX

Studer Revox America, Inc.
1425 Elm Hill Pike, Nashville, TN 37210 (615) 254-5651
Offices: Los Angeles (213) 780-4234 / New York (212) 255-4462
Canada: Studer Revox Canada, Ltd.



Circle 30 on Reader Service Card

Introducing a present

Once you go through a recording session with the new ATR-124 24-channel recorder by Ampex, you'll want to go through another. Because with each new session you'll discover something new you can do. Things that you can only do with a recorder that's full of features of the future.

ATR-124 gives you the unheard of: Time on your hands.
Which means you can use that time to give clients more of what they're paying for—your creative skills. With the ATR-124 microprocessor-based control system, you can pre-program what you want to do ahead of time so you won't waste studio time setting things up. When their time starts, you're ready to record by touching a single recall button.

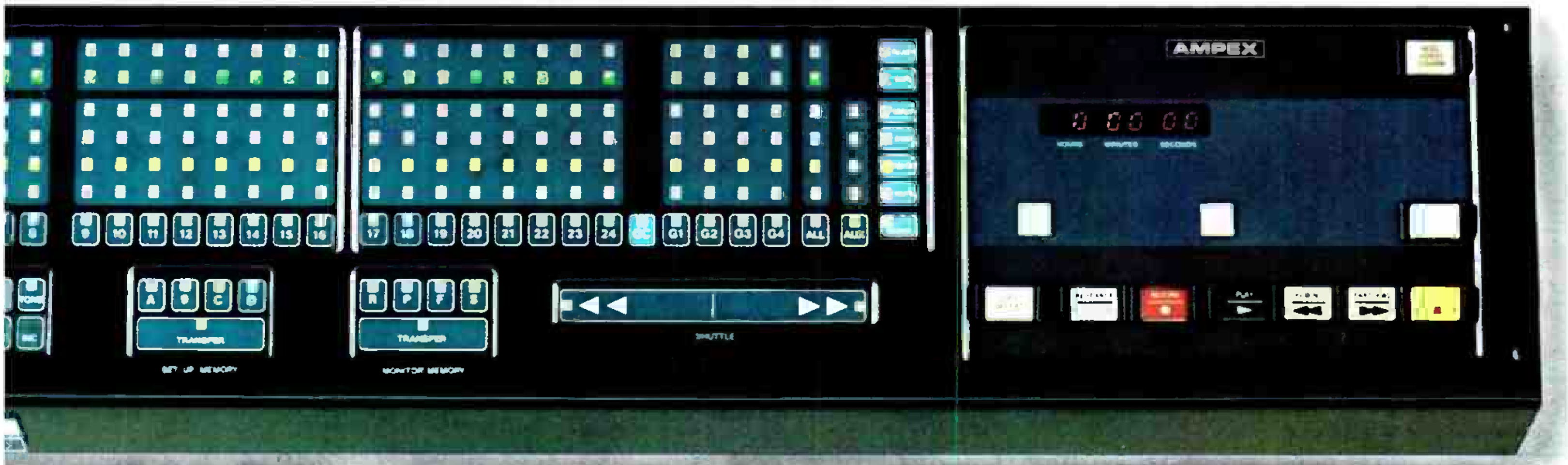
ATR-124 also lets you duplicate a technique you may have used earlier in the session without having to rethink what you did. Just touch the memory button and it'll all come back to you. ATR-124 lets you rehearse what you've got in mind, without recording it, to make sure what you've got in mind is right. Tape can be manipulated faster which means you'll get the sound you want sooner. And the chance to try something "a little different." All because of the speed and accuracy that ATR-124 puts at your fingertips.

ATR-124 doesn't take away your creativity, it adds to it.
The less time spent setting up, correcting, and redoing, the more time spent creating. And when you add features that help you create to the ones that help you save time, you've got one very potent piece of audio machinery. Take the control panel for instance. It's like nothing you've ever seen. Pushpads linked to a microprocessor give you a new level of creative flexibility. Program a setup, then change it. Then change it back, all with a single fingertip.

A repeatable, variable speed oscillator for pitch correction and special effects is built in. In addition

rom the future: ATR-124.

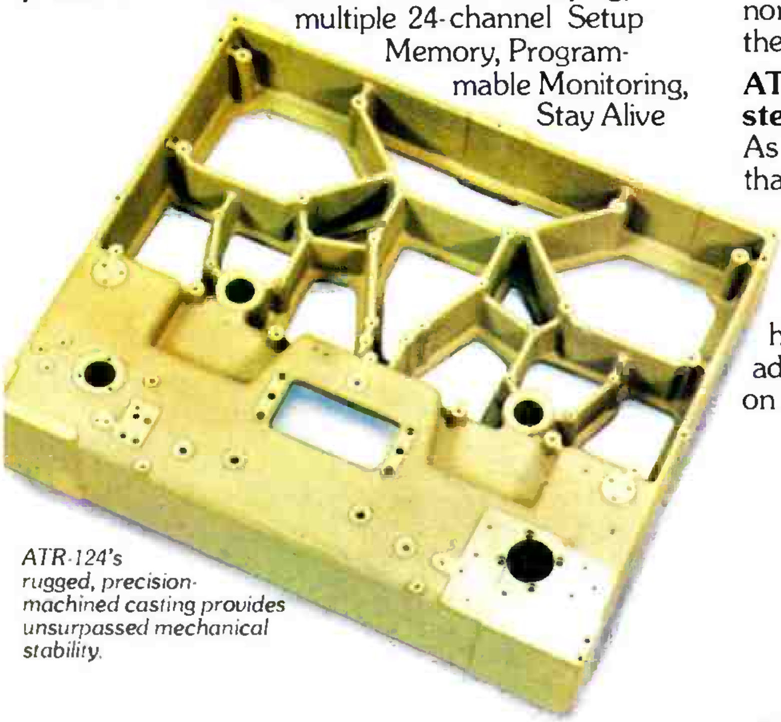
to the standard output, there is an optional auxiliary output with each channel that enhances flexibility. So don't think that ATR-124 is going to



ATR-124's Control Panel. Speed and accuracy at your fingertips.

replace anything that you do. On the contrary, it's going to improve the skills you have, if not help you develop some new ones.

ATR-124 picks up where ATR-100 leaves off.
It's only natural that the people who brought you the ATR-100 should be the ones to bring you something better. ATR-124 offers you 24 channels instead of 4. You also get many new and exclusive features. The kind that have set Ampex apart from the crowd for the last 30 years. Features like balanced, transformerless inputs and outputs; a patented flux gate record head; 16" reel capability; input and output signal bus for setup alignment; membrane switch setup panel; fingertip-operated shuttle speed control; and microprocessor-based synthesized Varispeed -50% to +200% in .1% steps or in ¼ tone steps. ATR-124 also features microprocessor-based control of Channel Grouping, multiple 24-channel Setup Memory, Programmable Monitoring, Stay Alive Memory, and Record Mode diagnostics. The point is this: If you like the ATR-100, you're going to love working with the ATR-124.



ATR-124's rugged, precision-machined casting provides unsurpassed mechanical stability.

ATR-124 options.
As impressive as the ATR-124 itself.
With the addition of a built-in Multi-Point Search-To-Cue (MPSTC), you can rehearse edits and control five tape-time actuated events and be compatible with SMPTE time code. Separately controlled auxiliary output amplifiers with each channel provide simultaneous monitoring of normal and sync playback as well as all other monitoring modes. A roll-around remote control unit can also be added to the ATR-124 which contains all control features normally found on the main unit.

ATR-124's Multi-Point Search-To-Cue (MPSTC). Provides 100 cue locations.

ATR-124. Your next step is to experience it firsthand.
As you scan the points we've covered, remember that you're scanning just a small portion of ATR-124's story. We haven't even begun to discuss the accessibility of key components for easy servicing and minimal downtime, or the features we've built in to give you greatly improved tape handling. To find out more, write to us at the address shown below. We'll send you a brochure on ATR-124, our latest audio effort.

ATR-124. Pure 24-Channel Gold From Ampex.

AMPEX

Listen to the future

Audio Video Systems Division
401 Broadway, Redwood City, California 94063 415/367-2011

Circle 31 on Reader Service Card

BOB KATZ

Achieving Depth Perception in Recording

Achieving accurate depth perception takes a working knowledge of acoustic principles and the principles of diffusion— plus *a good ear.*

THE RECORDING ENGINEER should be both an artist and a scientist about his work. When he makes a new recording which he perceives as "better" (more pleasurable) than a previous recording, he should ask himself *why?* Perhaps the reason for the "better" recording was a changed microphone distance from the sound source. The engineer should then try to consider whether the proximity effect, the Fletcher-Munson effect, or some other documented acoustic principle entered into his "better" recording. Then, of course, he will take a tape measure and carefully measure the distance of the mikes from the source in order to duplicate the sound next time. Of course not! (I had to say that for fear that many of you would take the previous statement seriously.) A more realistic approach is to quantify his findings as best he can and then internalize them so he can continue with the black art known as Recording Engineering.

This article will show how several well-known acoustic principles, including the Haas effect, and the principles of diffusion, can aid us in the task of achieving accurate depth perception in our artistic recordings. Remember, however, that the ear is the ultimate judge of all recording quality; no matter how much we try to quantify, there will remain some mystery over why one recording gives us pleasure, and another does not.

Bob Katz is a freelance recording engineer working out of NYC.

Part One: The Perception of Depth

At first thought, it may seem that depth perception is simply a matter of the ratio of direct to reverberant sound in a recording. On the contrary, it is a much more involved acoustic process. Our binaural hearing apparatus is largely responsible for the perception of depth. But recording engineers were concerned with achieving depth even in the days of monophonic sound. In the monophonic days, *in general,* halls for orchestral recording were much deader than those of today. The dead acoustic characteristics of Studio 8H at NBC are clearly audible on the early Toscanini/NBC Symphony recordings. That same studio is used today in television, and its short reverberation time can be heard on TV's Saturday Night Live, at least until the boom microphone moves out to considerable distance from the performers, when the apparent reverberation increases. Hey, come to think of it, TV is monophonic sound, isn't it? (At least today, in the U.S.)

Why is it that monophonic recording and dead rooms seem to go well together? The answer is involved in two principles that work hand in hand: 1) *The masking principle* and 2) *The Haas effect.*

THE MASKING PRINCIPLE

The masking principle says that a louder sound will tend to cover (mask) a softer sound, especially if the two sounds lie in the same frequency range. If these two sounds happen to be the direct sound from a musical instrument and the reverberation from that instrument, then the initial reverberation can appear to be covered by the direct sound. When the direct sound ceases, the reverberant hangover is finally perceived.

In real rooms (as opposed to rooms reproduced over speakers), our two ears sense the reverberation as coming diffusely from all around us, and the direct sound as having a distinct single location. Thus, in real rooms, the masking effect is somewhat reduced by the ears' ability to sense direction.

In monophonic reproduction, the reverberation is reproduced from the same source speaker as the direct sound, and so we may perceive the room as deader than it really is, because of the masking effect. Furthermore, if we choose a recording hall that is very live, then the reverberation will tend to intrude on our perception of the direct sound, since both will be reproduced from the same location—the single speaker.

This is the ultimate explanation for the incompatibility of many stereophonic recordings with monophonic reproduction. The larger amount of reverberation tolerable in stereo becomes less acceptable in mono due to the masking effect.

THE HAAS EFFECT...

The Haas effect says that, in general, echos occurring within approximately 40 milliseconds of the direct sound become fused with the direct sound. We say that the echo becomes "one" with the direct sound, and only a loudness enhancement occurs.

A corollary to the Haas effect says that fusion (and loudness enhancement) will occur even if the closely-timed echo comes from a different direction than the original source. However, the brain will continue to recognize (binaurally) the location of the original sound as the proper direction of the source. The Haas effect allows nearby echos (up to approximately 40 ms. delay) to enhance an original sound *without* confusing its directionality.

...AND ITS RELATIONSHIP TO REAL ROOMS

We may say that these shorter echos which occur in a natural environment (from nearby walls and floor) are *correlated with* the original sound, as they have a direct relationship. The longer reverberation is *uncorrelated;* it is what we call the *ambience* of a room. Most deader recording studios have little or no ambient field, and the very deadest studios have only a few perceptible early reflections to support and enhance the original sound.

In a good stereo recording, the early correlated room reflections are captured with their correct placement; they support the original sound, help us locate the sound source as to distance and do not interfere with left-right orientation. The later uncorrelated reflections, which we call *reverberation*, naturally contribute to the perception of distance, but because they are uncorrelated with the original source the reverberation does not help us locate the original source in space. This fact explains why the multitrack mixing engineer discovers that adding artificial reverberation to a dry, single-miked instrument may deteriorate the sense of location of that instrument. If the recording engineer uses stereophonic miking techniques *and* a liver room instead, capturing early reflections on two tracks of the multitrack, the remix engineer will have an easier time adding artificial reverberation convincingly.

A BRIEF SUMMARY

In short: Depth is perceived binaurally. Monophonic reproduction provides a very limited sense of depth perception due to the masking effect. Early reflections (up to 40 ms. delay) carry most of the distance and location information in a stereophonic recording. Later, diffused reflections are the reverberation; they give information about the size of the room and some information about the distance of the source, yet do not help us discern its direction.

AIR ABSORPTION

Before we leave this topic there is one last contributor to the sense of distance in a natural acoustic environment, and that is the absorption qualities of air. As the distance to a sound source increases, the apparent high frequency response is reduced. This is another tool which the recording engineer uses to change distance. An interesting experiment is to alter a treble control while playing back a good orchestral recording. Notice how the apparent front-to-back depth of the orchestra changes considerably as you manipulate the high frequencies.

Part Two: Recording Techniques to Achieve Front-To-Back Depth

MINIMALIST TECHNIQUES

In the December 1979 db, Bruce Bartlett's "Stereo Microphone Technique" is a compendium of minimalist microphone techniques. All of the methods described (from coincident pair to widely-spaced pair or trio) are applicable to this discussion of front-to-back depth. Later, I will make a few observations on how the choice of a particular minimalist technique affects the perception of depth.

Referring to FIGURE 1, a musical group is shown in a hall cross section. Various microphone positions are indicated by letters A-F.

Microphones *A* are located very close to the front of the orchestra. As a result, the ratio of *A*'s distance from the back compared to the front is very large. (The back is about ten times the distance of the front in this picture.) Consequently, the front of the orchestra will be much louder in comparison to the rear. The rear instruments, at least because of extreme level differences, will seem farther back, and front-to-back depth will be exaggerated. There is much to be said, however, in favor of mike position *A*, since the conductor usually stands there, and he purposely places the softer instruments (strings) in the front, and the louder (brass and percussion) in the back, somewhat compensating for the level discrepancy due to location. Also, the radiation characteristics of the horns of trumpets and trombones help them to overcome distance. These instruments frequently sound closer than other instruments located at the same physical distance from the microphone.

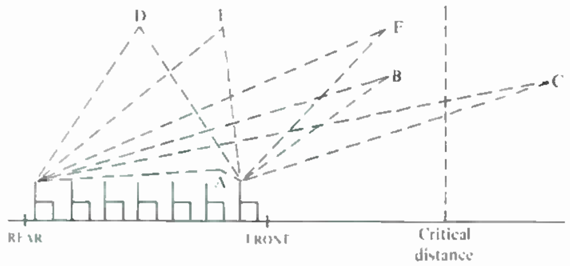
The other contribution to front-to-back depth is the larger ratio of reflected to direct sound for the back instruments. Relative reverberation contributes to the sense of front-to-back depth at mike position *A* or any other position.

The farther back we move into the hall, the smaller the ratio of back-to-front distance becomes and the softer the front of the orchestra sounds compared to the level of the back. At position *B*, the brass and percussion are only two times the distance from the mikes that the strings are. This (according to theory) makes the back of the orchestra 6 dB down compared to the front. Acoustics of the hall greatly change any decibel number quoted, since the more reverberant the hall, the less severe the level changes as we move farther back.

For example, in position *C*, the microphones are positioned beyond the critical distance[1] in the hall. As soon as we pass the critical distance, further changes in level of front orchestra compared to back become *much* less apparent to us. If the front of the orchestra seems too loud to us at position *B*, moving to *C* will not solve the problem. At position *C* we will probably hear about the same front-to-back relationship as at position *B*, except the entire orchestra will seem more distant and move overall room reverberation will be heard.

THE DIMENSION OF HEIGHT

The dimension of height allows us to change the front-to-back perspective practically independently of the reverberation perceived. At position *D*, there will be no front-to-back depth perceived, since the mikes are directly over the center of the orchestra, equidistant from front or back. Position *E* is the same distance from the orchestra as position *A*, but being much higher, the relative back-to-front ratio is much less. At *E* we may find the ideal depth perspective and a good level balance between the front and rear instruments. If even less front-to-back depth is desired, then *F* may be the solution, although with more overall reverberation and at a greater distance. Or we can try a position higher than *E*, with less reverb.



A hall cross section showing various microphone positions.

DIRECTIVITY OF MUSICAL INSTRUMENTS

Frequently the higher up we move in the hall, the more high frequencies we perceive, especially from the strings. This is because the high frequencies of many instruments (particularly violins and violas) radiate upward rather than forward. The high frequency factor adds more complexity to the problem, since it has been noted that treble response affects the apparent distance of a source. Note that when the mike moves past the critical distance in the hall, a listener may not hear significant changes in high frequency response when height is changed.

The recording engineer should be aware of how *all* the above factors affect the depth picture so he can make an intelligent decision as to which mike position to try next. Or else he will find himself in the position of the 100 monkeys seated at 100 typewriters. Sooner or later they will manage to type all the works of Shakespeare....Similarly, sooner or later, left entirely up to chance, any recording engineer will find that ideal mike position. Hopefully, *you* will know it when you've found it.

ADDING AMBIENCE...AND OTHER UGLY THINGS (or) THROWING THE MINIMALIST OUT THE WINDOW

The engineer/producer often desires a sense of warmth, ambience, or even distance which is not being achieved by the mike position, although the position is ideal for relative front-to-back orchestral balance and depth. In this case, moving the mikes back into the reverberant field cannot be the solution. Another case for increased ambience is when the hall itself is not the best hall for recording. In either case, trucking the entire ensemble to another hall may be tempting, but is not always the most practical solution.

Here, the engineer may decide to put up additional mikes to capture the hall ambience. If he has read his Burroughs and countless others on the evils of *acoustic phase cancellation*, he knows that adding more mikes is a sin. (This is of course why churches make the most ideal recording venues....The confessional is near enough to save the recording engineer a long trip which would otherwise pull him away from his main job of sinning.)

The fact is that acoustic phase cancellation does *not* occur when the extra mikes are placed purely in the reverberant field, for the reverberant field is uncorrelated with the direct sound. The problem, of course, is knowing where the reverberant field is located. Proper application of Burroughs' 3-to-1 rule will minimize acoustic phase cancellation. So will careful listening. The ambience mikes should be far enough back in the hall, *and* the hall must be reverberant enough so that when these mikes are mixed into the program, no deterioration in frequency response is heard, just an added warmth and increased reverberation character. If the ambience mikes are all the way back in the hall and a deterioration of frequency response is heard, then the hall is too dead—because the so-called reverberation actually contains correlated reflections which cause acoustic phase cancellation, also known as a *comb filter effect*. The engineer should instead cancel using the ambience mikes, and either try moving his main mikes further back, changing their directional pattern, or even a different hall than the one he shouldn't have chosen in the first place.

Assuming the added ambience consists of reverberation, which is uncorrelated with the direct source, then theoretically an artificial reverberation chamber should accomplish similar results to those obtained with ambience microphones. The answer is a qualified *yes*, assuming the artificial reverberation chamber sounds good and, consonant with the sound of the original recording hall.

What happens to the depth and distance picture of the orchestra as the ambience is added? In general, the front-to-back depth of the orchestra remains the same or increases minimally, but the apparent overall distance increases as more reverberation is mixed in. The change in depth may not be linear for the whole orchestra since the instruments with more dominant high frequencies may seem to remain closer even with added reverberation.

Part Three: The Influence of Hall Characteristics on Recorded Front-To-Back Depth

LIVE HALLS

In general, the more reverberant the hall, the farther back the rear of the orchestra will seem, given a fixed microphone distance. I am intimately familiar with one problem hall whose reverberation is much greater in the upper bass frequency region, particularly around 150 to 300 Hz.

A string quartet usually places the cello in the back. Since that instrument is very rich in the upper bass region, in this hall the cello always sounds farther away from the mikes than the second violin, which is located at his right. Strangely enough, a concert-goer in this hall does not notice the extra sonic distance because his strong visual sense locates the cello easily and does not allow him to notice an incongruity. When he closes his eyes, however, the astute listener notices that, yes, the cello sounds farther back than it looks!

It is therefore rather difficult to get a proper depth picture with microphones in this problem hall. Depth seems to increase almost logarithmically when low frequency instruments are placed only a few feet away from the mikes. It is especially difficult to record a piano quintet there because the low end of the piano excites the room so much as to make it seem too distant and hard to locate spatially. The problem is aggravated especially when the piano is on half-stick, cutting down the high frequency definition of the instrument.

The miking solution I choose for this problem is a compromise: close mike the piano, and mix this with a panning position that is the same as the piano's virtual image arriving from the main pair of front mikes. I can only add a small portion of this close mike before the apparent level of the piano is taken above the balance a listener would hear in the hall. The close mike does help to solidify the image and locate the piano somewhat. It gives the listener a little more direct sound on which to focus. More on multi-mike techniques shortly.

VERY DEAD ROOMS

What happens when minimalist techniques are applied to recording ensembles in dead studios? Not much. My observations are that the purist coincident pair and other two-mike techniques which generally sound superior to me are not so obviously superior in a dead studio. A recent experiment was made with a horn overdub (sax and brass section) on multi-track tape. The experiment took place at the facilities of the Institute of Audio Research, New York City, where I teach a recording course. Students at I.A.R. do have a bit more time to experiment with "exotic" miking choices than they would in the speedy outside world. I suggested that they try listening to a coincident pair placed so as to pick up a natural image of the horns. It seemed to us that *in this dead room* there were no significant differences between the sound of this "minimalist"

pair, and six "multiple-mono" close up mikes! The close mikes were, of course, carefully equalized, leveled and panned from left to right. This was certainly a surprising discovery to me and it points out the importance of good hall acoustics on a musical sound.

Normally, it would seem that close multiple miking kills all sense of depth, and it generally does. But in the dead room, the more-distantly-placed coincident pair did not pick up a significant number of room reflections to provide a reasonable or even pleasing depth feeling. When the horns were monitored and mixed with the other tracks on the multitrack, the rhythm section effectively masked whatever room ambience that might have been perceived through the coincident pair. The differences between the two miking techniques then sounded insignificant to me. In fact, while this conclusion may sound like heresy to some people out there, I even found that the balance control flexibility provided by the multiple close mikes made them the superior choice in this room.

THE LEFT-TO-RIGHT PICTURE

Bartlett reports that the further apart the microphone pair, the wider the stereo image of the ensemble. Instruments near the sides tend to pull more left or right. Center instruments tend to get wider and more diffuse in their image picture, harder to locate or focus spatially.

The technical reasons for this effect are tied in to the Haas effect for delays of under approximately 5 ms. vs. significantly longer delays. With very short delays between two spatially located surfaces, the left-to-right image location becomes ambiguous. Fusion does not occur. A listener can experiment with this effect by mistuning the azimuth on a two-track machine and playing a mono tape over a well-focused stereo speaker system. When the azimuth is correct, the center image should be tight and defined. When the azimuth is mistuned, the center image should get wider and acoustically out of focus. Similar problems can (and do) occur with the mike-to-mike time delays always present in spaced-pair techniques.

THE FRONT-TO-BACK PICTURE

I have found that when the left-to-right picture is getting wider (due to an increase in intermike distance), the depth picture also appears to increase, especially in the center. This is probably due to the extra wide and diffuse center image. For example, the front line of a chorus will no longer seem straight. Instead, it appears to be on an arc bowing away from the listener in the middle. If soloists are placed at the left and right sides of this chorus instead of in the middle, a rather pleasant and workable artificial depth effect will occur. You should not overrule spaced-pair techniques on the basis of theory alone. In fact, two well-known practitioners of the spaced mike technique (Bob Fine, who used three equally-spaced omnidirectionals, and David Hancock, who uses two figure-8 mikes spaced a few feet apart) have produced orchestral recordings which in my opinion equal the best ever made in terms of their imaging, depth and focusing accuracy.

MULTIPLE MIKING TECHNIQUES

I have described how multiple close mikes destroy the depth picture; in general I stand behind that statement. But soloists do exist in orchestras, and for many reasons, they are not always positioned in front of the group. The engineer should make an effort to relocate the soloist closer to the front microphones when looking for a natural depth picture. But when the soloist cannot be moved, plays too softly, or when hall acoustics make him sound too far back, then a close mike or mikes must be used.

Check for frequency response problems when the close mike is mixed in. As noted before, the live hall is more forgiving and, if little or no acoustic phase cancellation occurs, the close mike can be used. The close mike (not surprisingly) will appear to bring the solo instrument closer to the listener. If this practice is not overdone, the effect is not a problem as long as musical balance is maintained, and the close mike levels *are not changed* during the performance. We've all heard recordings made with this disconcerting practice. To me, they make the soloist sound like he must be on roller skates because his distance changes at the whim of the recording engineer/producer!

WHAT ABOUT DELAY MIXING?

At first thought, adding a delay to the close mike seems attractive. While this delay will synchronize the *direct* sound of that instrument with the *direct* sound of that instrument arriving at the front mikes, the single delay line cannot effectively simulate the other delays of the multiple early room reflections surrounding the soloist. The multiple early reflections arrive at the distant mikes and contribute to direction and depth. They *do not* arrive at the close mike with significant amplitude compared to the direct sound entering the close mike. Therefore, while delay mixing may help, it is not a panacea.

STEREO MIKING A SOLOIST

When the close solo mikes are a properly placed stereo pair *and* the hall is not too dead, the depth image will seem more natural than one obtained with a single solo mike. However, the advantages of stereo miking a soloist decrease in recordings made in *dead* rooms, for reasons mentioned earlier. (Blame it all on the Haas effect. He's very dead anyway.)

INFLUENCE OF THE CONTROL ROOM ENVIRONMENT ON THE DEPTH PICTURE

This is the last topic. Many engineers may at this point say, "what is this matter of depth? I've never noticed it in my control room."! But chances are, if you've continued to read this far, you have noticed a degree of depth perception in your recordings and want to know how to make them sound "better." (That word "better" has inspired more fights in the audio industry than I care to admit. Unfortunately, this article will be no exception.)

In my opinion, there are now more good speakers available for home stereo use that are capable of producing an accurate depth image than are found in today's so-called "state-of-the-art" recording studios! But that situation is rapidly changing. In control rooms of the past, the acoustic design engineer's decisions for efficiency, high power handling, flat frequency response, wide bandwidth across the console width, and other speaker attributes, have been made at the expense of a tight center image and accurate depth and focus. I believe that this trend is changing considerably with the recent advent of the LEDE™ control room and phase-coherent speaker systems.

Before you book your next studio gig, play a well-recorded orchestral tape that you are familiar with (in the control room). If you find the flutes and clarinets sound like they are four feet wide and nowhere in space, then I suggest you book another studio. Happy listening, and happy depth perception! ■

FOOTNOTE

[1] "Critical Distance" is a term used in Sound Reinforcement work; I believe it was first popularized or coined by Don Davis. It refers to the point in the hall where the direct and reverberant sound are at equal level.

BIBLIOGRAPHY

1. Bartlett, Bruce: "Stereo Microphone Technique," db, December 1979.
2. Burroughs, Lou: "Microphones: Design and Application," Sagamore Publishing, 1974.
3. Madsen, E. Roerbaek: "Extraction of Ambiance Information from Ordinary Recordings," Journal of the Audio Engineering Society, 1970 October. Contains an excellent explanation of the Haas effect as well as a description of a four-speaker ambiance extraction system that must be heard to be believed. (It is excellent).
4. Davis, et al: various recent publications on the LEDE concept and on TDS from db and RE/P magazines.

db New Products & Services

PROFESSIONAL MONITOR SYSTEM

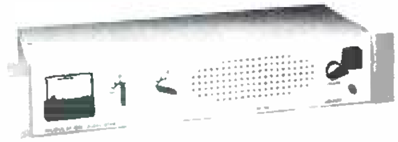
• The Sentry 100 Professional Monitor System employs a tweeter capable of handling 25 watts RMS of input power, while faithfully reproducing the program material with response out to 18 kHz and uniform dispersion, 120 degrees at 5 kHz. Accidental high-frequency blasts from tape head contact in rewind/fast forward mode no longer results in destruction of the tweeter. The low-frequency section is an 8-in. direct radiator woofer installed in an optimally vented enclosure with fourth-order Butterworth tuning. The Sentry 100 has a uniform polar response so that the engineer hears the same sound 30 degrees off-axis as he does directly in front of the system, and a high-frequency control that offers boost as well as cut.
Mfr: Electro-Voice
Circle 40 on Reader Service Card

MONITOR MIXING CONSOLE

• The Audy Series 2000M Monitor Mixing Console provides 16 inputs (stackable to 32) with separate output mixes that permit control of up to six independent monitor sends. Using high speed, low noise IC op-amp technology, it minimizes transient and slewing-induced intermodulation distortion. A dual LED system assures proper adjustments of input attenuation switches and maintains 25 dB of headroom throughout. Other standard features include: input and output channel patching; EQ in/out switch for each input mix control; individual channel muting; talkback; 6 auxiliary inputs; headphone monitoring with solo priority system; high resolution, 20-segment LED bar-graph meters; phantom power, and work lamp socket.
Mfr: Audy Instruments, Inc.
Price: $6,995.00
Circle 41 on Reader Service Card

AUDIO MONITOR

• The AM200 audio monitor has been designed to high impedance bridge six individual program circuits. Each selected channel can be monitored for audio level by use of a front panel VU meter and range select switch that covers a 60 dB dynamics range. Each channel can also be subjectively monitored with the use of a front panel speaker system. In addition, an independent remote speaker output is provided.
Mfr: Modulation Associates, Inc.
Circle 42 on Reader Service Card



CONDENSER MICROPHONE

• The SM85 hand-held vocal microphone is especially suitable for applications requiring wide frequency response, low distortion characteristics, very low RF susceptibility and reliable operation over a wide range of temperature and humidity extremes. It features an integral multistage pop filter, a midrange presence peak, extended high-frequency response, a controlled low-frequency rolloff and an internal shock mount for reduced handling noise. The SM85 is designed for simplex (phantom) powering from an external supply or directly from sound reinforcement, broadcast or recording equipment. It operates over a voltage range of 11 to 52V dc, covering both DIN standard 45 596 simplex voltages of 12 and 48 volts and the proposed 24-volt standard. Frequency response is 50 to 15,000 Hz.
Mfr: Shure Brothers Inc.
Price: $231.00
Circle 43 on Reader Service Card

AUDIO PROCESSING SYSTEM

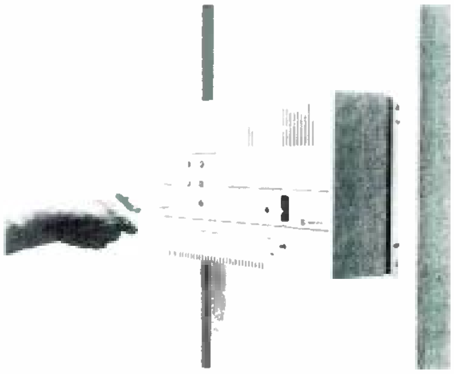
• The Emph' a Sizer, a new audio processing system from ATI, combines the functions of a program controlled input gate, a switchable four band parametric equalizer and a wide range, low distortion compressor-limiter. Simplified controls in a compact, RF protected package make the Emph' a Sizer ideal for use in the studio as a DJ mike processor with switchable, presettable equalizers to tailor the Emph' a Sizer for each announcer. Equally useful in the recording studio and in sound reinforcement, the Emph' a Sizer accepts direct microphone or line level inputs and provides both low and high level balanced outputs to +24 dBm.
Mfr: Audio Technologies, Inc.
Price: $1,395.00
Circle 44 on Reader Service Card

AMPLIFIER SERIES

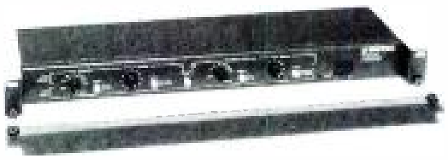
• The TU-A Series of telephone paging/utility public address amplifiers are available with 100, 60 or 35 watt ratings, designated by model numbers TU-100A, TU-60A and TU-35A, respectively. Each screwdriver-installed, wall- or 19-in.-rack-mounting model has a peak-reading LED to indicate when the amplifier is driven into clipping, an illuminated power switch and dual circuit protection from shorts or overloads. A strip of input/output no-solder, screw terminal connections is easily accessible on the front panel and protected by a screwdown cover. Three 600-ohm telephone lines can be connected to the unit. Two of their inputs are transformer-isolated. Channel 1 is a combination 150-ohm balanced microphone/600-ohm line input. Two channels provide high-impedance auxiliary inputs and one input has precedence capability as standard. Output facilities include 25V, 25V center-tapped and 70V lines and 16 ohms, balanced or unbalanced. The TU-A amplifiers deliver rated power from 100 to 5,000 Hz with less than 5 percent total harmonic distortion. Frequency response is ±2 dB from 70 to 12,000 Hz.

Mfr: LSI Bogen

Circle 45 on Reader Service Card



ELECTRONIC CROSSOVER



• The EX-18 is a new electronic crossover that can be used as a stereo two-way or as a mono three-way crossover. A mode switch allows easy conversion from two-way to three-way operation with no external patching required. A single knob on each channel adjusts the crossover frequency from 100 Hz to 1600 Hz. For very high frequency operation, a 10-X multiplier switch provides continuous adjustment between 1 kHz and 16 kHz. The EX-18 uses 18 dB/octave Butterworth filters to clearly separate the high frequencies from the low frequencies. A high-frequency phase inverting switch provides a quick method for optimizing the phase of the speaker system.

Mfr: E-V/TAPCO

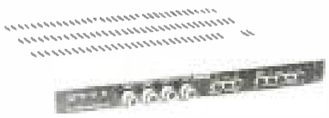
Circle 47 on Reader Service Card

VIDEO CHARACTER GENERATOR

• The VCG-750 Series Video Character Generator is a versatile, microprocessor-based SMPTE/EBU time code instrument which reads and displays longitudinally recorded serial SMPTE/EBU time code and/or Vertical Interval Time Code. The VCG-750 reads standard time code from video tape or any other SMPTE/EBU source at speeds ranging from zero to 60 times normal play. From this time code input, the new unit extracts tape time and user bit data and outputs this data as video characters which may be burned into or superimposed over the source video for display. A unique feature of the VCG-750 is that, in the Vertical Time Code Mode, the unit is capable of reading time code at 30 times play speed, forward and reverse. An Auto Code Source Mode can also be specified, in which switchover from serial to Vertical Interval Time Code occurs automatically at tape speeds below 1/5 play, or whenever Serial Time Code drops out or becomes invalid. Compatible with international video standards, the VCG-750 accommodates 24, 25, and 30 frame-per-second drop frame and non-drop frame SMPTE/EBU time code inputs.

Mfr: EECO Incorporated

Circle 46 on Reader Service Card



POWER AMPLIFIER

• The Model A60 Power Amplifier is designed for professional applications requiring audiophile-quality performance. The compact chassis houses a modular, fully complementary amplifier capable of 4000 watts rated power dissipation. Functional features include LED Fault, Signal Present and Thermal indicators; automatic balanced/unbalanced inputs, and automatic mono input. Total harmonic distortion is less than .05 percent from 20 Hz to 20 kHz. The exterior chassis is constructed of 5¼-in.-high heavy gauge steel. The power transformer and associated power supply components are housed in a separate internal chassis, with the transformer mounted very near the front panel.
Mfr: Phase Linear Corporation
Circle 48 on Reader Service Card

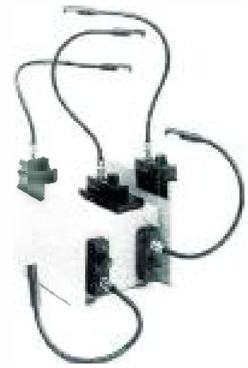
MUSIC LIBRARY

• The Series VI is a new three album package of production music from Soper Sound. The 59-selection series is being sold, as is all the music in the three year old library's catalogue, on a buy-out basis that entitles the buyer unlimited use in all productions, any media, any sized market for an unlimited time. Series VI contains a new metronome tempo guide, an extended instrumentation guide and a list of key words called descriptors. Other changes within Series VI include a new policy of 30 and 60 second versions of the appropriate themes. Series VI contains all original music, recorded in 24-track studios. The orchestration ranges from small ensembles through complete horn and string sections.
Mfr: Soper Sound Music Library
Circle 49 on Reader Service Card

AUDIO ADAPTERS

• Switchcraft, Inc. has recently introduced Series 399M audio adapters for its line of "Q-G" audio connectors. The four adapters are available in the Switchcraft Public Address Audio Adapter Kit, giving the audio technician the four most commonly needed PA connector/adapters in one package. Included are a 3-pin male Q-G to 3-contact male threaded-coupling microphone connector; a 3-pin male Q-G to 4-contact female threaded-coupling microphone connector, and a 3-pin male Q-G to 4-pin male threaded-coupling microphone connector. These adapters are designed to mate with all Switchcraft 3-pin male Q-G connectors and similar types. On the other end, they mate with Amphenol 3- or 4-contact male or female threaded-coupling microphone connectors.
Mfr: Switchcraft, Inc.
Circle 50 on Reader Service Card

MOUNTING ACCESSORIES



• The Littlelites line from Custom Audio has been expanded to include three new mounting accessories: the Adjustable Mounting Clip (CL), the Weighted Base (WB), and the Plastic Snap Mount (SM). The CL, for use in temporarily clamping an L-1 or L-2 lamp onto music stands, console sideplates, clipboards, etc., adjusts to clamp thickness of 1/16-in. to ¾-in. The WB allows free standing operation of any L-1 or L-2, where application requires a moveable lamp. The SM is used for semi-permanent placement of an L-1 or L-2 on equipment. It enables fast placement of the lamp directly over equipment in use, and easy removal for storage, with minimal effect on the equipment.
Mfr: Custom Audio Electronics, Inc.
Circle 51 on Reader Service Card

TRANSFORMERLESS AMPLIFIER

• The Model 5008 Hybrid Dual Balanced Transformerless Amplifier is designed to operate from bipolar supplies of from 12 to 20 volts. The specifications of the 5008 make it suitable for use in audio frequency circuits with stringent requirements. It has equivalent Input Noise of −110 dB; THD of 0.05%, 20 Hz to 20 kHz, at +20 dBm output; Frequency Response of ±0.25 dB, 20 Hz to 20 kHz, and a slew rate of 9V per millisecond. Crosstalk from one channel to the other channel is −80 dB at 20 kilohertz. The 5008 will drive a 600 ohm output, resistive. The Model 5008 package may be used in a wide variety of applications by simple external addition of resistors or interconnections. Some typical applications include: dual bridging or matching differential inputs in a single ended outputs with gains to 14 dB and inverting summing amplifiers.
Mfr: Modular Audio Products
Price: $52.00
Circle 52 on Reader Service Card

MOSFET AUDIO SWITCHER

• The MAP Model 4088 MOSFET Audio Switcher provides for individual remote double throw switching of eight line level inputs. A small jumper plug changes input-output and interconnections of the two sections. The card may be used as a Pan-Direct switcher with On-Off for two stereo outputs, or as an On-Off switcher for a monaural input to eight outputs directly or through 20K ohm summing resistors, or as a Pre-Post-Off switch for four mono inputs. A MAP Model 5008 BTA module in the input circuit serves as a buffer amplifier to insure isolation and nonloading of the inputs, and provides up to 14 dB of gain to make up for fader setting loss when the card is used as a channel routing switcher in console applications. The switching logic is TTL compatible with permissible switching voltages from +4.5 volts to +15 volts.

Mfr: Modular Audio Products
Circle 56 on Reader Service Card

PRO MICROPHONE STANDS

• The Omni Series angulated microphone stands are available in floor model for the stand-up performer, lectern height for table-top use, and with optional horizontal boom extension. They depart from the single-position right angle stands, and are supplied in a glare-free chrome finish. In addition to the traditional 90 degrees setting, the Omni Series is equipped with an adjustable fulcrum offering the advantage of controlled microphone-proximity, and the option of 75 or 105 degree angular tube positioning. Omni Series stands and boom extensions are equipped with the standard ⅝-in. number 27 thread suitable for all U.S. microphone holders and accessories.

Mfr: Atlas Sound
Circle 57 on Reader Service Card

POLYPHONIC SEQUENCER

• The DSX Digital Polyphonic Sequencer has a three thousand note memory, expandable to a six thousand note capacity. With features such as editing, overdubbing, sixteen voice capability, and the ability to merge sequences in any order, the DSX allows control of composition previously unavailable in an inexpensive sequencer. Other features include eight independently controlled control voltage outputs and eight gate outputs, each with user selectable positive- or negative-going outputs. Real-time programming, as well as a single step mode, and synchronization to tape for multi-tracking are other important capabilities.

Mfr: Oberheim Electronics
Price: $1,700.00
Circle 58 on Reader Service Card

GRAPHIC EQUALIZERS

• The 1651A active graphic equalizer is a single channel equalizer with 10 minimum phase shift, active band rejection filter sections. Center detented slide controls provide ±12 dB boost/cut at I.S.O. preferred 1-octave frequencies (31.5-16 kHz). The 1652A is a stereo graphic equalizer with the same features as the 1651A for each of two channels. Both the 1651A and 1652A incorporate a continuously variable high-pass filter with 18 dB/octave roll-off, and a user-selectable low-pass filter with 6 dB roll-off at 12.5 kHz. The 1653A is a fourth-generation EQ device that delivers 1/3-octave accuracy across a wide range of industrial and professional applications. 29 minimum phase shift, active band rejection filter sections use center detented slide controls to provide ±12 dB boost/cut. Continuously variable high-pass and low-pass filters provide roll-off at 18 dB/octave from off position to 20 through 160 Hz (high-pass), and from off position to 5 kHz through 20 kHz (low-pass). Filters in all units are parallel summed so that failure of one section will not affect remaining filter section operation.

Mfr: Altec Lansing
Circle 59 on Reader Service Card

PATCHBAY

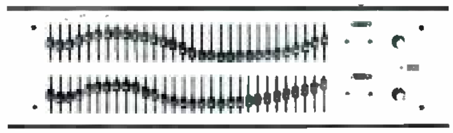
• The Model Patch-32 is a patchbay designed for interconnection of unbalanced audio devices. The rear panel of the Patch-32 accepts up to 32 unbalanced audio signals via ¼-in. phone jacks (or optional RCA jacks), and routes them to 32 ¼-in. front panel jacks. For user convenience, the front panel, top row jacks are normalized to the front panel bottom row jacks such that automatic connections between two devices can be made without a patchcord between upper and lower rows. All the jacks in the Patch-32 are isolated from the Patch-32's chassis to prevent possible ground loops between devices.

Mfr: Symetrix, Inc.
Price: $149.00
Circle 60 on Reader Service Card

MICROCOMPOSER

• The MC-4 MicroComposer is a computer specially designed for writing and performing musical compositions. Up to four pairs of voices can be programmed from either a synthesizer keyboard and/or the MC-4's own calculator style keyboard. Performances can include programmed pitch, rhythm, programmed dynamics, programmed tempo and more. Many advanced functions are available including syncing to tape for building multitrack performances of any complexity. The composition/performance data in the MC-4 memory can be stored and retrieved for later use with the self-contained tape memory interface or through the optional MTR-100 Digital Data Cassette recorder, which includes automated search for specific data tracks and exceptionally fast, reliable data storage and retrieval. The MC-4 is available with up to 48K byte of RAM memory for the musicians use, allowing up to 11,500 notes of composition/performance at one time. The MC-4 also includes 16K byte of ROM, an internal program that is specially designed to offer nearly limitless musical flexibility and functional options.
Mfr: Roland Corp US
Price: $3,295.00
Circle 62 on Reader Service Card

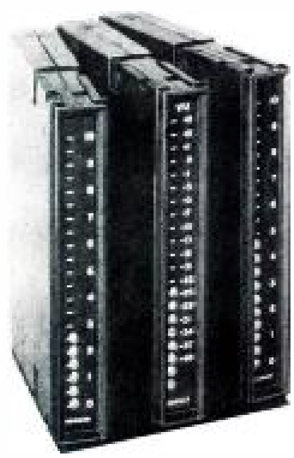
GRAPHIC EQUALIZER



• The DN30/30 dual-channel 1/3 octave graphic equalizer has two completely discrete channels, with 30 bands of control each, centered on standard I.S.O. frequencies. The range control is switchable for each channel, ±6 dB or ±12 dB. In addition, each channel contains a 30 Hz, 18 dB/octave subsonic filter, switchable in or out. The DN30/30 incorporates an earth-lift switch and a system-bypass facility for power interruptions. The equalizer can also be fitted with internal active crossover circuit cards in either a bi-amp or tri-amp configuration. Slope and center frequency are user selectable.
Mfr: Klark-Teknik Electronics Inc.
Price: $1,450.00
Circle 65 on Reader Service Card

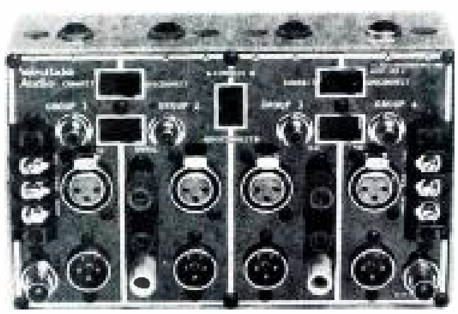
ANALOG BARGRAPH

• The APM 20 is a new solid state analog bargraph indicator featuring a 3-in., 20 element bargraph. APM meters provide a low cost, reliable alternative to mechanical meters and are available in many standard voltmeter and ammeter ranges. Options include offset span, differential input, reduced response time and either single or dual setpoint controls. Meters meet ANSI 39.1 shock, temperature and humidity requirements. Display brightness is controllable with an external potentiometer.
Mfr: Bowmar/ALI Inc.
Price: $60.00
Circle 63 on Reader Service Card

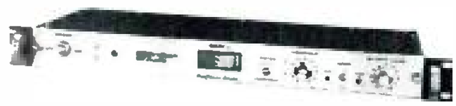


STEREO, CONNECTOR MULT AND INTERFACE BOX

• The CMB-2 functions as an interface adapter box, signal multing box, signal switching box, and as a troubleshooting aid. Completely passive in design, the CMB-2 accommodates male and female XLR, Phone (TRS), BNC, Banana and Terminal Strip connections. Through use of switch selection, the connectors can be isolated into a maximum of four sections or they can function as one continuous, twenty six connector, three conductor mult.
Mfr: Westlake Audio
Circle 64 on Reader Service Card

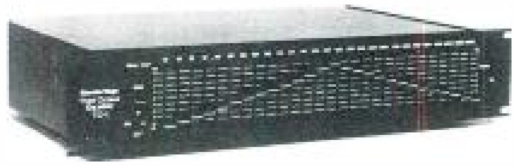


COMPRESSOR-LIMITER



• The Model 663CL Compressor-Limiter is a compact, rack-mounting device with a self-contained power supply. Plug-in PC motherboard construction allows the entire electronics section to be removed, serviced, or replaced from the front. The front panel controls of the 663CL include: compression threshold; output level; VU meter switchable to output or compression, and a power on-off switch/circuit breaker. Input and output are 600 ohms, transformer coupled, balanced, floating, and capable of handling up to +27 dBm level before clipping. Compression is 20 dB maximum, with a ratio of approximately 2.5 to 1. The compression control circuits are accessible externally, allowing two or more units to be interconnected for stereo or multi-channel operation.
Mfr: ProTech Audio
Price: $495.00
Circle 66 on Reader Service Card

THIRD OCTAVE EQUALIZER



• The basic architecture of the TO-1 third octave equalizer is a single mono unit. The front panel contains all of the operating controls and indicators. There are 27 band controls, each calibrated for 12 dB boost and cut. The band centers occur on ISO standard frequencies running from 40 Hz through 16,000 Hz. Located to the left of the sliders are a clipping indicator, a rotary gain control and an EQ in/out switch. The EQ in/out switch determines whether the equalization circuitry is active. The gain control has a control range from "off" to +6 dB of flat gain. The gain setting is unaffected by the EQ in/out switch position. On the rear panel are the input and output connectors that allow the TO-1 to interface with a variety of sound systems. The TO-1 provides both balanced and unbalanced inputs as well as outputs. The TO-1 is equipped with an active filter network called a GIC (Generalized Impedance Converter) using operational ICs. The GIC circuitry works by simulating an RLC bandpass filter.
Mfr: Cerwin-Vega
Price: $600.00
Circle 53 on Reader Service Card

16-TRACK CONSOLE

• The M-16, a new 16-track console, can be configured with 16 or 24 input positions. There are eight main program mixing busses with submaster faders, eight main board outputs, two independent stereo mixing busses, four auxiliary mixing busses and 16 meters switchable to read buss or external source. The M-16 has stereo solo in-place for input, monitor and effects returns. All inputs have pre-fader-listen capability. During remix, the solo logic permits instant comparisons between effects send/receive. The M-16 also features 4-band, 8-knob parametric (sweep type) equalizers that may be switch bypassed, and three filters (two high-pass, one low-pass). Faders are conductive plastic with a 100 mm throw.
Mfr: TEAC Corporation
Price: $12,900.00
Circle 54 on Reader Service Card

AIR CONDENSER MICROPHONES

• "The Alternative" is a cost effective precision air condenser microphone family for OSHA and other ANSI Type I acoustic measurement and monitoring applications. The ACO Pacific 7012 (free field ½-in.) and the 7013 (pressure response ½-in.) are direct replacements for the Bruel and Kjaer 4133 and 4134 respectively.
Mfr: ACO Pacific, Inc.
Price: $450.00
Circle 55 on Reader Service Card

New Literature

CABLE ASSEMBLY GUIDE

• A new 28-page illustrated catalog and specification guide from Belden Corp.'s Interconnect Systems Operation describes design-it-yourself molded cable assemblies for interfacing data systems, peripherals, instrumentation, and controls. Among the shielded and unshielded standard product lines covered are 9- and 37-position RS-449; 15- and 50-position RS-422A and RS-423A; 25-position RS-232C; and 24-position IEEE 488 GPIB. Also described are Series 10 and 20 finger phone plug assemblies for instrumentation and control circuits; Series 50 miniature and subminiature phone plug assemblies for audio connection in miniaturized instrument, automation, and control systems; and Series 60 phono plug assemblies for instrumentation and audio control applications. The design-it-yourself format of the catalog enables specifiers to select from a wide variety of component configurations, including 100-percent shielded designs, to construct an assembly that combines electrical and mechanical features to meet specific application requirements. **Mfr: Belden Corp., Interconnect Systems Operation, 105 Wolfpack Rd., Gastonia, N.C. 28052.**

FREE SMPTE CATALOG

• The 1981 catalog of SMPTE publications has recently been published by the Society of Motion Picture and Television Engineers (SMPTE). The new free catalog features descriptions of SMPTE publications including the newly-published book *Television Technology in the 80's*. Other books that are described are the three-volume series *Digital Video—Volumes 1, 2, & 3*. Several books in the motion-picture field are listed including the best-selling *Motion-Picture Projection and Theatre Presentation Manual* and the classic *Special Effects in Motion Pictures: Some Methods for Producing Mechanical Special Effects*. **Mfr: SMPTE, 862 Scarsdale Ave., Scarsdale, NY 10583.**

MICROPHONE BROCHURE

• Gotham Audio Corporation has announced the availability of a new brochure for Neumann fet 80 Condenser Microphones. This six page, four color brochure features the complete line of microphones including the new KMR 82 Shotgun, USM 69 Stereo microphone and the U 89 with its revolutionary new capsule design. Descriptions of each microphone as well as complete technical specifications are included. Also featured are Gotham's special products including colored windscreens, the N 80 power supply and colored cables. **Mfr: Gotham Audio Corporation, 741 Washington St., New York, NY 10014.**

ANALOG PANEL METERS BULLETIN

• Bowmar's new analog panel meters expressly designed for high visibility and readability are described in a bulletin available from Bowmar. All available models are described along with specifications and outline drawings. **Mfr: Bowmar/ALI Inc., 531 Main Street, Acton, Ma. 01720.**

db Classified

Closing date is the fifteenth of the second month preceding the date of issue.
Send copies to: Classified Ad Dept.
db THE SOUND ENGINEERING MAGAZINE
1120 Old Country Road, Plainview, New York 11803

Minimum order accepted: $25.00
Rates: $1.00 a word
Boxed Ads: $40.00 per column inch
db Box Number: $8.50 for wording "Dept. XX," etc.
Plus $1.50 to cover postage

Frequency Discounts: 6 times, 15%; 12 times, 30%

ALL CLASSIFIED ADS MUST BE PREPAID.

FOR SALE

SCULLY MODEL 100-16 TRACK S/N 200. One owner since new, plenty of spare parts, mint condition. $12,500.00, includes full remote control. **Contact Bob or Bubba, (512) 690-8888.**

REELS AND BOXES 5" and 7" large and small hubs, heavy duty white boxes. **W-M Sales, 1118 Dula Circle, Duncanville, Texas 75116 (214) 296-2773.**

FOR SALE: EMT 250 Digital Reverb Unit, mint condition. $17,500. **RPM Sound Studios, 12 E. 12th St., New York, NY 10003. (212) 242-2100.**

AMPEX SPARE PARTS; technical support; updating kits, for discontinued professional audio models; available from **VIF International, Box 1555, Mountain View, Ca. 94042. (408) 739-9740.**

TECHNICS, BGW, EVENTIDE, AKG, Scully, and many more IN STOCK. FOR IMMEDIATE DELIVERY: **UAR Professional Systems, 8535 Fairhaven, San Antonio, TX 78229. (512) 690-8888.**

FREE CATALOG & AUDIO APPLICATIONS

CONSOLES
KITS & WIRED
AMPLIFIERS
MIC., EQ, ACN, LINE, TAPE, DISC, POWER
OSCILLATORS
AUDIO, TAPE BIAS
POWER SUPPLIES

OPAMP LABS INC.
1033 N. SYCAMORE AVE.
LOS ANGELES, CA. 90038
(213) 934-3566

REMANUFACTURED ORIGINAL equipment capstan motors for all Ampex and Scully direct drive recorders, priced at $250., available for immediate delivery from **VIF International, PO Box 1555, Mtn. View, CA 94042, phone (408) 739-9740.**

DISC CUTTING SYSTEM, Westrex 2B head rest of system stereo, Scully Lathe 2 track Ampex, Limiter, Eqs., Altec/McIntosh Monitors, priced to sell. **Soundwave, 50 W. 57 St., N.Y.C. 10019 (212) 582-6320.**

AGFA MASTERTAPE AND CASSETTES. Super prices. Example: ¼" x 2400' bulk = 9.82 and C-60 for $.86 (case quantities) send for wholesale price list. **Solid Sound, Inc., Box 7611, Ann Arbor, Mich. 48107 (313) 662-0667.**

ELECTRO-VOICE SENTRY Studio Monitors, Otari Recorders, dbx Systems, AB Systems Amplifiers, Harmon/Kardon hi/fi products. Best prices—immediate shipment. **East: (305) 462-1976, West: (213) 243-1168.**

FOR SALE: PENTAGON SR-4250-12 reel master/one (1) reel slave duplicator with electronics. ¼-track configuration. Less than 2 years old. Good condition. Make offer. Contact: **Mikie Baker, P.O. Box 400329, Dallas, TX 75240, (214) 233-2900.**

TASCAM, BGW, JBL, EV, Nikko, Technics, Onkyo, Hafler, Audio Research, Conrad Johnson, dbx, etc. **P. K. Audio, 4773 Convention, Baton Rouge, LA 70806. (504) 924-1001.**

---

101 RECORDING SECRETS MOST ENGINEERS WON'T TELL, $7.95 **Tunetronics, P.O. Box 55, Edgewater, N.J. 07020.**

ORBAN. All products in stock. FOR IMMEDIATE DELIVERY. **UAR Professional Systems, 8535 Fairhaven, San Antonio, TX 78229. 512-690-8888.**

AMPEX, OTARI, SCULLY—In stock, all major professional lines; top dollar trade-ins; write or call for prices. **Professional Audio Video Corporation, 384 Grand Street, Paterson, New Jersey 07505. (201) 523-3333.**

THE LIBRARY...Sound effects recorded in STEREO using Dolby throughout. Over 350 effects on ten discs. $100.00. Write **The Library, P.O. Box 18145, Denver, Colo. 80218.**

ACOUSTIC TEST INSTRUMENTS
USE TO ADJUST EQUALIZERS, OPTIMIZE SPEAKER PLACEMENT, ETC.

Complete Test Set	$399
Calibrated Sound Level Meter	$169
Fractional Octave Pink Noise Generator	$299
Handbook on Acoustic Testing	$6

Free broch HALL ENGINEERING, Dept E4
P O Box 506, Martinsville, NJ 08836
(201) 647-0377

EMPLOYMENT

SERVICES

People/Places/Happenings

• **Maxell Corporation of America,** 60 Oxford Drive, Moonachie, NJ, a manufacturer of high quality magnetic media, and **Harris Corporation,** Broadcast Product Div., Quincy, IL, a major manufacturer of equipment for the radio and television industry, have become sustaining members of the **Society of Motion Picture and Television Engineers** (SMPTE). By joining the SMPTE, Maxell and Harris join more than 190 sustaining member companies and more than 9,000 individual members in the motion picture and television industries. The SMPTE is well known for its work in standardization, the publication of the monthly *SMPTE Journal* and the sponsorship of technical conferences and equipment exhibits.

• **Fanfare Audio Enterprises, Inc.,** the touring sound and production company, have announced several major developments. Fanfare opened a new branch location in Orlando, Florida on July 1, 1981 and is expanding its services over a wide range of fields. In addition to major tour rentals, Fanfare now has available P.A. systems for rent out of its new Orlando location, in addition to its Jackson, Michigan base. Other new areas include an expanded sales program, custom installations and convention services. According to **Martin Preece,** president, Fanfare's new space efficient four-way all-in-one cabinet will be ready for touring in late August.

• **John Stachowiak,** veteran recording engineer and specialist in the cutting of master disks, has recently opened **Disk Master Studio.** Disk Master is the only studio in the United States equipped with all the components of the newest disk mastering system made by **Neumann,** the world-renowned producer of microphones and other top-quality equipment. Stachowiak has 15 years of professional experience as a recording engineer, including six years specializing in disk mastering, which began during his four-and-a-half-year employment as a staff engineer at **Sound Recorders Studio** in Hollywood. As a free-lance engineer, Stachowiak has been associated with several Los Angeles-area studios, including **The Record Plant, Crystal, Allen Zentz,** and **Filmways-Heider (also, Heider, San Francisco), and with** Columbia Studios in London. Artists for whom he has mastered disks include the **Beatles, Kenny Rogers, Neil Diamond, Richard Harris, Herb Alpert** and **Judy Collins.**

• **Sam Borgerson** has been appointed advertising and public relations manager of **Studer/Revox America,** according to an announcement made by the company's president, **Bruno Hochstrasser.** Borgerson will direct all advertising and related activities from the firm's principal office in Nashville. Borgerson is already known to many in the recording industry through his byline articles in several professional trade magazines. He has previous experience as a concert sound mixer, retail hi-fi salesman, and advertising copywriter. In 1977-78, he served as a Sales Coordinator in the Revox Sales Department.

• **Charles P. Coovert** has been appointed general manager of the audio products group of the **Ampex Audio-Video Systems Division.** The announcement was made by **Donald V. Kleffman,** vice-president-general manager of the division. Coovert will be responsible for the development, manufacturing and marketing of all Ampex audio products in his new position. He succeeds **Lee Cochran,** who resigned.

Coovert was most recently senior product manager in the video recorder group of AVSD. He was previously operational systems manager in the controller's department for the division and inventory manager for the International division. Coovert was associated with **Precision Instruments** in Palo Alto, California, before joining Ampex in 1971.

• **Audio-Video Resources,** San Francisco, CA, celebrated its announcement of new expansion plans with a champagne Open House, which was attended by representatives of the Bay Area video, audio, and advertising industry. A.V.R., with newly remodeled facilities, including a new 24 track audio sweetening facility, opened its doors for voice-overs and mix-downs. Audio-Video Resources also announced the formation of a new division—AVR Productions, which is capable of video and audio production from conception to final dub. AVR Productions is already at work on projects for **Ford Motor Company** and **Natomas.** Recent additions to the staff of Audio-Video Resources have been **Rich Poggioli** from **C.C.R.** and **United Nations Television** in New York, as video facilities manager, and **Jayne Morris** from **WNEW-TV,** as the chief video editor. The most recent addition has been **Jim Cassedy,** who joins AVR as video engineer from **United Artists,** in New York.

• **Trutone Records Disk Mastering Labs** has announced the expansion and complete renovation of their disk mastering labs and offices located just outside of New York City, at 163 Terrace Street, Haworth, N.J. Construction is underway for two new mastering rooms. The unique architectural/acoustical design concept was developed and created by the team of **Maurice Wasserman/Al Fierstein.** The main cutting room will be equipped with their newly acquired **Neumann** VMS-70 lathe, SX-74 head, SAL-74 rack and Neumann mastering console. The second room will be equipped with the **Scully/Westrex** system featuring the new **Capps Generation II** computer and lathe control.

• **Dr. Harry F. Olson,** retired vice president, **RCA Laboratories,** has been awarded the **Gold Medal** of the **Acoustical Society of America.** The citation states, "For his innovative and lasting contributions in acoustic transducers, sound-reproduction, electronic music and speech synthesis and his service to the Society." Besides this latest honor, Dr. Olson has received many awards from several scientific and engineering societies. Dr. Olson has written eleven books and holds more than 100 patents in acoustics and sound reproduction. He is a past President of the Acoustical Society of America and the **Audio Engineering Society** and a member of the **National Academy of Sciences.**

• Describing it as, "One of the largest single orders ever received by our International Division," **Altec Lansing** announced it will supply components for all major public address systems in the New King Khaled International Airport. Under construction at the Saudi Arabian capital of Riyadh, the multi-billion dollar facility will utilize Altec and **University Sound** products for a system including 11,000 speakers and producing 85,000 watts of audio output. As Altec Professional Sound Contractors for the massive project, **Acromedia Corporation,** Culver City, CA, was awarded the contract, subsequently presenting the order to Altec Lansing officials at the company's Anaheim, CA, headquarters.

• **Larry Weston,** president of **Edcor,** Irvine, CA, announced the appointment of **James E. Morrison** to vice president, marketing. Jim Morrison had previously served as vice president of sales for **Altec Corporation** and sales manager for **University Sound.**

MASTERS OF THE STUDIO.

There's as much magic in the mixing board as there is in the keyboard.

That's why, when we award the Ampex Golden Reel, it goes to both the recording artist *and* the recording studio. Together they provide the magic that turns a reel of recording tape into an outstanding creative achievement.

The Ampex Golden Reel Award honors those achievements that were mastered on Ampex professional recording tape. They've earned a place in the ranks of the world's most successful recorded albums and singles*.

Along with the Award, we also present $1,000 to a charitable organization. Since we started the Golden Reel Awards three years ago, there have been over 200 recipients, and more than $200,000 donated on their behalf.

Congratulations to all of them. The masters on both sides of the microphone.

AMPEX

Ampex Corporation, Magnetic Tape Division, 401 Broadway, Redwood City, California 94063, (415) 367-3889.

 Golden Reel Winners as of 1/81

*RIAA Certified Gold.

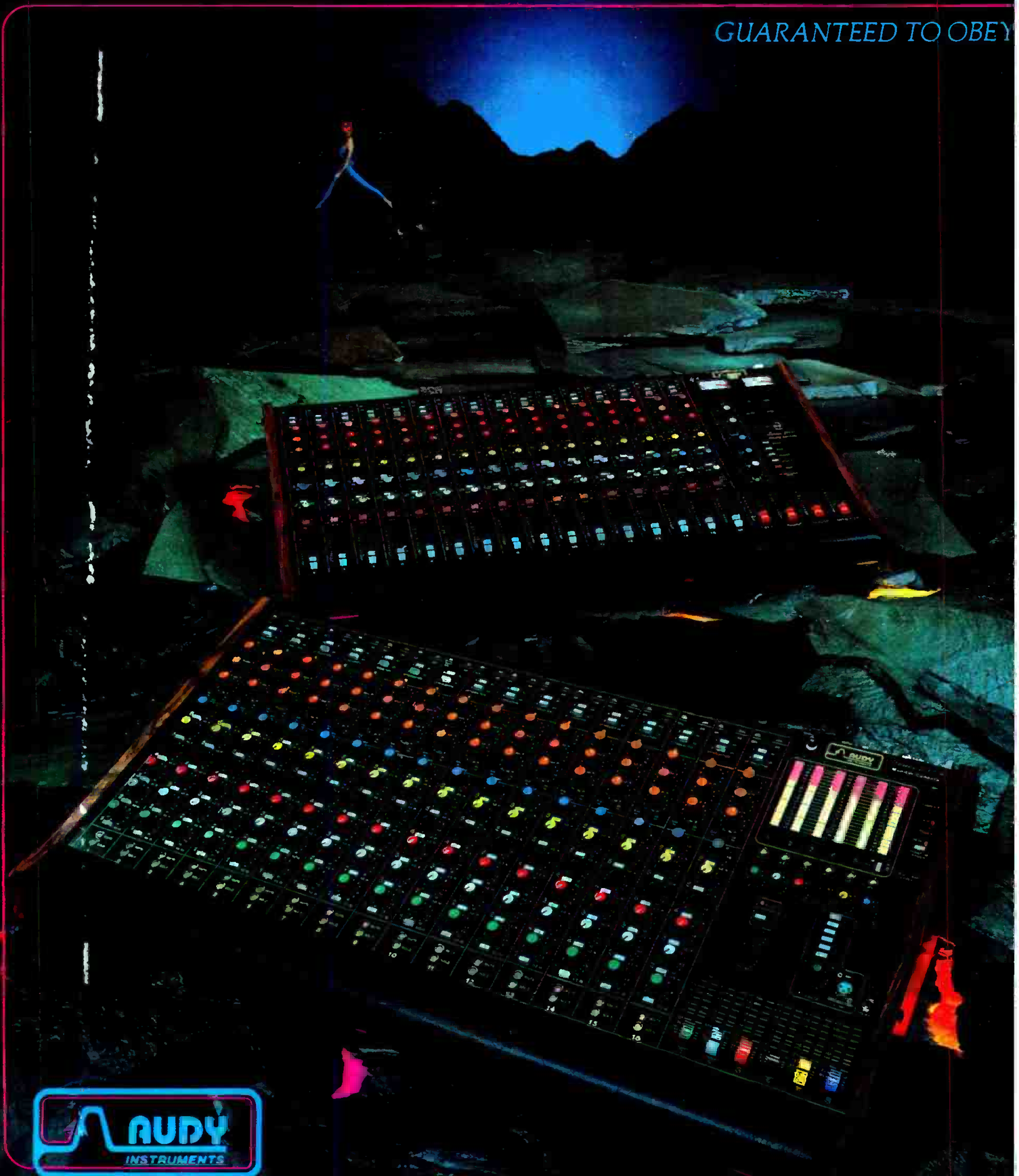
Of all the mixer manufacturers in the industry, why is Audy the only one to offer a full two year warranty? Simple. We can back it up.

We can state unequivocably that the Audy Series 2000 will obey whatever commands your performance requirements dictate.

Because Audy is more than a manufacturer. We're our own toughest customer. A professional concert sound company for top artists throughout North America, Audy conceived and designed the Series 2000 for one simple reason: no other mixer was good enough.

And thousands of concerts later, we're still convinced that the Series 2000 is the master of its class.

The Audy Series 2000 and Series 2000M monitor consoles. Command and they will obey. We guarantee it.



For a free, full color brochure, write Audy Instruments, Inc., P.O. Box 2054, 35 Congress Street, Shetland Industrial Park, Salem, MA 01970. (617) 744-532(

Circle 12 on Reader Service Card